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(54) **INTONATION ADJUSTMENT IN TEXT-TO-SPEECH SYSTEMS**
INTONATIONSREGELUNG IN TEXT-ZU-SPRACHE-SYSTEMEN
REGLAGE DE L'INTONATION DANS DES SYSTEMES TEXTE-PAROLE

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[0001] The present invention relates to translating text in a computer system to synthesized speech; and more particularly to techniques used in such systems for control of intonation in synthesized speech.

[0002] In text-to speech systems, stored text in a computer is translated to synthesized speech. As can be appreciated, this kind of system would have wide spread application if it were of reasonable cost. For instance, a text-to-speech system could be used for reviewing electronic mail remotely across a telephone line, by causing the computer storing the electronic mail to synthesize speech representing the electronic mail. Also, such systems could be used for reading to people who are visually impaired. In the word processing context, text-to-speech systems might be used to assist in proofreading a large document.

[0003] However in prior art systems which have reasonable cost, the quality of the speech has been relatively poor making it uncomfortable to use or difficult to understand. In order to achieve good quality speech, prior art speech synthesis systems need specialized hardware which is very expensive, and/or a large amount of memory space in the computer system generating the sound.

[0004] Prior art systems which have addressed this problem are described in part in United States Patent No. 8,452,168, entitled COMPRESSION OF STORED WAVE FORMS FOR ARTIFICIAL SPEECH, invented by Sprague; and United States Patent No. 4,692,941, entitled REAL-TIME TEXT-TO-SPEECH CONVERSION SYSTEM, invented by Jacks, et al. Further background conceming speech synthesis may be found in United States Patent No. 4,384,169, entitled METHOD AND APPARATUS FOR SPEECH SYNTHESIZING, invented by Mozer, et al.

[0005] In text-to-speech systems, an algorithm reviews an input text string, and translates the words in the text string into a sequence of diphones which must be translated into synthesized speech. Also, text-to-speech systems analyze the text based on word type and context to generate intonation control used for adjusting the duration of the sounds and the pitch of the sounds involved in the speech.

[0006] Diphones consist of a unit of speech composed of the transition between one sound, or phoneme, and an adjacent sound, or phoneme. Diphones typically are encoded as a sequence of frames of sound data starting at the center of one phoneme and ending at the center of a neighboring phoneme. This preserves the transition between the sounds relatively well. The encoded diphones have a nominal pitch determined by the length of a pitch period in the encoded speech and a nominal duration determined by the number of pitch periods corresponding to a particular encoded sound. These nominal values must be adjusted to synthesize natural sounding speech.

[0007] Intonation control in such systems involves lengthening or shortening particular frames, or pitch periods, of speech data for pitch control, and inserting or deleting frames associated with particular sounds for duration control. Prior art systems have accomplished these modifications by relatively crude clipping and extrapolation on pitch period boundaries that introduce discontinuities in output speech data sequences. In some cases, these discontinuities may introduce audible clicks or other noise.

[0008] Notwithstanding the prior work in this area, the use of text-to-speech systems has not gained widespread acceptance. It is desireable therefore to provide a software only text-to-speech system which is portable to a wide variety of microcomputer platforms, and conserves memory space in such platforms for other uses, and performs intonation control with high quality.

[0009] The present invention provides a software-only real time text-to-speech system including intonation control which does not introduce discontinuities into output speech stream. The intonation control system adjusts the intonation of sounds represented by a sequence of frames having respective lengths of digital samples. It includes a means that receives intonation control signals and a buffer for storing frames in the sequence of sound data. The intonation control system is responsive to the intonation control signals for modifying a block of one or more frames in the sequence to generate a modified block. The modified block substantially preserves the continuity of the beginning and ending segments of the block with adjacent frames in the sequence. Thus, when the modified block is inserted in the sequence, no discontinuities are introduced and smooth intonation control is accomplished.

[0010] According to the invention, there is provided an apparatus for adjusting an intonation of a sound wherein the sound is specified by a sequence of frames each comprising a set of digital samples, the apparatus comprising: means for receiving a set of intonation control signals that indicate a pitch adjustment and a duration adjustment to the sound; a buffer that stores the sequence of frames; intonation control means that generates an intonation adjusted sequence of frames by accessing a block of one or more frames of the sequence of frames from the buffer and by generating a modified block in response to the intonation control signals and by inserting the modified block into the sequence of frames; characterized by comprising means for applying a first weighting function to the block emphasizing the beginning segment to generate a first vector and means for applying a second weighting function to the block emphasizing the ending segment to generate a second vector, and means for combining the first vector with the second vector to generate the modified block, such that the intonation control means minimizes a discontinuity between a beginning segment and an ending segment of the block and a pair of adjacent frames in the intonation adjusted sequence of frames.

[0011] According to one embodiment of the invention, the intonation control signals include pitch control signals which indicate an amount of adjustment of the nominal lengths of particular frames in the sequence. Also, the intonation control signal may include duration control signals which indicate an amount to reduce or increase the number of frames in the sequence corresponding to particular sounds.

[0012] The pitch adjustment means includes a pitch lowering module which increases the length N of a particular frame by amount of Δ samples. In this case, the block which is modified consists of the particular frame. A first weighting function is applied to the block in the buffer emphasizing the beginning segment to generate a first vector, and a second weighting function is applied to the block emphasizing the ending segment to generate a second vector. The first vector is combined with the second vector shifted by Δ samples to generate a modified block of length N + Δ.

[0013] A pitch raising module is included for decreasing the length N of a particular frame by amount Δ. In this case, the block stored in the buffer consists of the particular frame subject of pitch adjustment and the next frame in the sequence of length NR. A first weighting function is applied to the block emphasizing the beginning segment to generate a first vector, and a second weighting function is applied to the block emphasizing the ending segment to generate a second vector. The first vector is combined with the second vector shifted by Δ samples to generate a shortened frame, and the shortened frame is concatenated with the next frame to produce a modified block of length N-Δ + NR.

[0014] Duration control includes duration shortening modules and duration lengthening modules. In the duration shortening module, the duration control signals indicate an amount to reduce the number of frames in a sequence that correspond to a particular sound. In this case, the block stored in the buffer consists of two sequential frames of respective lengths NL and NR which correspond to a particular sound. A first weighting function is applied to the block emphasizing the beginning segment to generate a first vector, and a second weighting function is applied to the block emphasizing the ending segment to generate a second vector. The first and second vectors are combined to generate a modified block having the length either NL or the length NR.

[0015] The duration lengthening module is responsive to duration control signals which indicate an amount to increase the number of frames in the sequence which correspond to a particular sound. In this case, the block to be modified consists of left and right sequential frames of respective lengths NL and NR which correspond to the particular sound. A first weighting function is applied to the block emphasizing the beginning segment to generate a first vector. A second weighting function is applied to the block emphasizing the ending segment to generate a second vector. The first and second vectors are combined to generate a new frame for insertion in the sequence. The left frame, the new frame, and the right frame are concatenated to produce the modified block.

[0016] According to a preferred embodiment of the invention, the intonation control is explicitly applied to speech data, in a text-to-speech system. The text-to-speech system includes a module for translating text to a sequence of sound segment codes and intonation control signals. A decoder is coupled to the translator to produce sets of digital frames which represent sounds for the respective sound segment codes in the sequence. An intonation adjustment module as described above is included which is responsive to the translator, and to modify the outputs of the decoder to produce an intonation adjusted sequence of data. An audio transducer receives the intonation adjusted sequence to produce synthesized speech.

[0017] By modifying speech data to adjust the intonation without introducing discontinuities between frames of speech data, a much improved text-to-speech system is achieved. Furthermore, the present invention is well suited to real time application in a wide variety of standard microcomputer platforms, such as the Apple Macintosh class computers, DOS based computers, UNIX based computers, and the like. The system occupies a relatively small amount of system memory, and utilizes the relatively small amount of processor resources to achieve very high quality synthesized speech.

[0018] Other aspects and advantages of the present invention can be seen upon review of the figures, the detailed description of preferred embodiments, given by way of example only, and the claims which follow.

[0019] Fig. 1 is a block diagram of a generic hardware platform incorporating the text-to-speech system of the present invention.

[0020] Fig. 2 is a flow chart illustrating the basic text-to-speech routine according to the present invention.

[0021] Fig. 3 illustrates the format of diphone records according to one embodiment of the present invention.

[0022] Fig. 4 is a flow chart illustrating an encoder for speech data according to the present invention.

[0023] Fig. 5 is a graph discussed in reference to the estimation of pitch filter parameters in the encoder of Fig. 4.

[0024] Fig. 6 is a flow chart illustrating the full search used in the encoder of Fig. 4.

[0025] Fig. 7 is a flow chart illustrating a decoder for speech data according to the present invention.

[0026] Fig. 8 is a flow chart illustrating a technique for blending the beginning and ending of adjacent diphone records.

[0027] Fig. 9 consists of a set of graphs referred to in explanation of the blending technique of Fig. 8.

[0028] Fig. 10 is a graph illustrating a typical pitch versus time diagram for a sequence of frames of speech data.

[0029] Fig. 11 is a flow chart illustrating a technique for increasing the pitch period of a particular frame.

[0030] Fig. 12 is a set of graphs referred to in explanation of the technique of Fig. 11.

[0031] Fig. 13 is a flow chart illustrating a technique for decreasing the pitch period of a particular frame.

[0032] Fig. 14 is a set of graphs referred to in explanation of the technique of Fig. 13.

[0033] Fig. 15 is a flow chart illustrating a technique for inserting a pitch period between two frames in a sequence.

[0034] Fig. 16 is a set of graphs referred to in explanation of the technique of Fig. 15.

[0035] Fig. 17 is a flow chart illustrating a technique for deleting a pitch period in a sequence of frames.

[0036] Fig. 18 is a set of graphs referred to in explanation of the technique of Fig. 17.

[0037] A detailed description of preferred embodiments of the present invention is provided with reference to the figures. Figs. 1 and 2 provide an overview of a system incorporating the present invention. Fig. 3 illustrates the basic manner in which diphone records are stored according to the present invention. Figs. 4-6 illustrate encoding methods based on vector quantization of the present invention. Fig. 7 illustrates a decoding algorithm according to the present invention.

[0038] Figs. 8 and 9 illustrate a preferred technique for blending the beginning and ending of adjacent diphone records. Figs. 10-18 illustrate the techniques for controlling the pitch and duration of sounds in the text-to-speech system.

I. System Overview (Figs. 1-3)

[0039] Fig. 1 illustrates a basic microcomputer platform incorporating a text-to-speech system based on vector quantization according to the present invention. The platform includes a central processing unit 10 coupled to a host system bus 11. A keyboard 12 or other text input device is provided in the system. Also, a display system 13 is coupled to the host system bus. The host system also includes a non-volatile storage system such as a disk drive 14. Further, the system includes host memory 15. The host memory includes text-to-speech (TTS) code, including encoded voice tables, buffers, and other host memory. The text-to-speech code is used to generate speech data for supply to an audio output module 16 which includes a speaker 17.

[0040] According to the present invention, the encoded voice tables include a TTS dictionary which is used to translate text to a string of diphones. Also included is a diphone table which translates the diphones to identified strings of quantization vectors. A quantization vector table is used for decoding the sound segment codes of the diphone table into the speech data for audio output. Also, the system may include a vector quantization table for encoding which is loaded into the host memory 15 when necessary. Also, the text-to-speech code in the instruction memory includes an intonation control module which preserves the continuity of encoded speech, while providing sophisticated pitch and duration control.

[0041] The platform illustrated in Fig. 1 represents any generic microcomputer system, including a Macintosh based system, an DOS based system, a UNIX based system or other types of microcomputers. The text-to-speech code and encoded voice tables according to the present invention for decoding occupy a relatively small amount of host memory 15. For instance, a text-to-speech decoding system according to the present invention may be implemented which occupies less than 640 kilobytes of main memory, and yet produces high quality, natural sounding synthesized speech.

[0042] The basic algorithm executed by the text-to-speech code is illustrated in Fig. 2. The system first receives the input text (block 20). The input text is translated to diphone strings using the TTS dictionary (block 21). At the same time, the input text is analyzed to generate intonation control data, to control the pitch and duration of the diphones making up the speech (block 22). The intonation control signals in the preferred system may be produced for instance as described in the related applications.

[0043] After the text has been translated to diphone strings, the diphone strings are decompressed to generate vector quantized data frames (block 23). After the vector quantized (VQ) data frames are produced, the beginnings and endings of adjacent diphones are blended to smooth any discontinuities (block 24). Next, the duration and pitch of the diphone VQ data frames are adjusted in response to the intonation control data (block 25 and 26). Finally, the speech data is supplied to the audio output system for real time speech production (block 27). For systems having sufficient processing power, an adaptive post filter may be applied to further improve the speech quality.

[0044] The TTS dictionary can be implemented using any one of a variety of techniques known in the art. According to the present invention, diphone records are implemented as shown in Fig. 3 in a highly compressed format.

[0045] As shown in Fig. 3, records for a left diphone 30 and a record for a right diphone 31 are shown. The record for the left diphone 30 includes a count 32 of the number NL of pitch periods in the diphone. Next, a pointer 33 is included which points to a table of length NL storing the number LP_i for each pitch period, i goes from 0 to NL-1 of pitch values for corresponding compressed frame records. Finally, pointer 34 is included to point to a table 36 of ML vector quantized compressed speech records, each having a fixed set length of encoded frame size related to nominal pitch of the encoded speech for the left diphone. The nominal pitch is based upon the average number of samples for a given pitch period for the speech data base.

[0046] A similar structure can be seen for the right diphone 31. Using vector quantization, a length of the compressed speech records is very short relative to the quality of the speech generated.

[0047] The format of the vector quantized speech records can be understood further with reference to the frame encoder routine and the frame decoder routine described below with reference to Figs. 4-7.

II. The Encoder/Decoder Routines (Figs. 4-7)

[0048] The encoder routin is illustrated in Fig. 4. The encoder accepts as input a frame s_n of speech data. In the preferred system, the speech samples are represented as 12 or 16 bit two's complement numbers, sampled at 22,252 Hz. This data is divided into non-overlapping frames s_n having a length of N, where N is referred to as the frame size. The value of N depends on the nominal pitch of the speech data. If the nominal pitch of the recorded speech is less than 165 samples (or 135 Hz), the value of N is chosen to be 96. Otherwise a frame size of 160 is used. The encoder transforms the N-point data sequence s_n into a byte stream of shorter length, which depends on the desired compression rate. For example, if N=160 and very high data compression is desired, the output byte stream can be as short as 12 eight bit bytes. A block diagram of the encoder is shown in Fig. 4.

[0049] Thus, the routine begins by accepting a frame s_n (block 50). To remove low frequency noise, such as DC or 60 Hz power line noise, and produce offset free speech data, signal s_n is passed through a high pass filter. A difference equation used in a preferred system to accomplish this is set out in Equation 1 for $0 \le n < N$.

$$x_n = s_n - s_{n-1} + 0.999 * x_{n-1} \qquad \text{Equation 1}$$

[0050] The value x_n is the "offset free" signal. The variables s_{-1} and x_{-1} are initialized to zero for each diphone and are subsequently updated using the relation of Equation 2.

$$x_{-1} = x_N \text{ and } s_{-1} = s_N \qquad \text{Equation 2}$$

[0051] This step can be referred to as offset compensation or DC removal (block 51).

[0052] In order to partially decorrelate the speech samples and the quantization noise, the sequence x_n is passed through a fixed first order linear prediction filter. The difference equation to accomplish this is set forth in Equation 3.

$$y_n = x_n - 0.875 * x_{n-1} \qquad \text{Equation 3}$$

[0053] The linear prediction filtering of Equation 3 produces a frame y_n (block 52). The filter parameter, which is equal to 0.875 in Equation 3, will have to be modified if a different speech sampling rate is used. The value of x_{-1} is initialized to zero for each diphone, but will be updated in the step of inverse linear prediction filtering (block 60) as described below.

[0054] It is possible to use a variety of filter types, including, for instance, an adaptive filter in which the filter parameters are dependent on the diphones to be encoded, or higher order filters.

[0055] The sequence y_n produced by Equation 3 is then utilized to determine an optimum pitch value, P_{opt}, and an associated gain factor, β. P_{opt} is computed using the functions $s_{xy}(P)$, $s_{xx}(P)$, $s_{yy}(P)$, and the coherence function Coh (P) defined by Equations 4, 5, 6 and 7 as set out below.

$$s_{xy}(P) = \sum_{n=0}^{N-1} y_n * PBUF_{P_{max} - P + n}$$

Equation 4

$$s_{xx}(P) = \sum_{n=0}^{N-1} y_n * y_n$$

Equation 5

$$s_{yy}(P) = \sum_{N=0}^{N-1} PBUF_{P_{max}} - P + n * PBUF_{P_{max}} - P + n$$

Equation 6

and

$$Coh(P) = s_{xy}(P) * s_{xy}(P) / (s_{xx}(P) * s_{yy}(P)) \qquad \text{Equation 7}$$

[0056] PBUF is a pitch buffer of size P_{max}, which is initialized to zero, and updated in the pitch buffer update block 59 as described below. P_{opt} is the value of P for which Coh(P) is maximum and $s_{xy}(P)$ is positive. The range of P considered depends on the nominal pitch of the speech being coded. The range is (96 to 350) if the frame size is equal to 96 and is (160 to 414) if the frame size is equal to 160. P_{max} is 350 if nominal pitch is less than 160 and is equal to 414 otherwise. The parameter P_{opt} can be represented using 8 bits.

[0057] The computation of P_{opt} can be understood with reference to Fig. 5. In Fig. 5, the buffer PBUF is represented by the sequence 100 and the frame y_n is represented by the sequence 101. In a segment of speech data in which the preceding frames are substantially equal to the frame y_n, PBUF and y_n will look as shown in Fig. 5. P_{opt} will have the value at point 102, where the vector y_n 101 matches as closely as possible a corresponding segment of similar length in PBUF 100.

[0058] The pitch filter gain parameter β is determined using the expression of Equation 8.

$$\beta = s_{xy}(P_{opt}) / s_{yy}(P_{opt}). \qquad \text{Equation 8}$$

[0059] β is quantized to four bits, so that the quantized value of β can range from 1/16 to 1, in steps of 1/16.

[0060] Next, a pitch filter is applied (block 54). The long term correlations in the pre-emphasized speech data y_n are removed using the relation of Equation 9.

$$r_n = y_n - \beta * PBUF_{P_{max} - P_{opt} + n}, \qquad 0 \leq n < N. \qquad \text{Equation 9}$$

[0061] This results in computation of a residual signal r_n.

[0062] Next, a scaling parameter G is generated using a block gain estimation routine (block 55). In order to increase the computational accuracy of the following stages of processing, the residual signal r_n is rescaled. The scaling parameter, G, is obtained by first determining the largest magnitude of the signal r_n and quantizing it using a 7-level quantizer. The parameter G can take one of the following 7 values: 256, 512, 1024, 2048, 4096, 8192, and 16384. The consequence of choosing these quantization levels is that the rescaling operation can be implemented using only shift operations.

[0063] Next the routine proceeds to residual coding using a full search vector quantization code (block 56). In order to code the residual signal r_n, the n point sequence r_n is divided into non-overlapping blocks of length M, where M is referred to as the "vector size". Thus, M sample blocks b_{ij} are created, where i is an index from zero to M-1 on the block number, and j is an index from zero to N/M-1 on the sample within the block. Each block may be defined as set out in Equation 10.

$$b_{ij} = r_{Mi+j}, \ (0 \leq i < N/M \text{ and } j \leq 0 < M) \qquad \text{Equation 10}$$

[0064] Each of these M sample blocks b_{ij} will be coded into an 8 bit number using vector quantization. The value of M depends on the desired compression ratio. For example, with M equal to 16, very high compression is achieved (i. e., 16 residual samples are coded using only 8 bits). However, the decoded speech quality can be perceived to be somewhat noisy with M = 16. On the other hand, with M = 2, the decompressed speech quality will be very close to that of uncompressed speech. However the length of the compressed speech records will be longer. In the preferred implementation, the value M can take values 2, 4, 8, and 16.

[0065] The vector quantization is performed as shown in Fig. 6. Thus, for all blocks b_{ij} a sequence of quantization vectors is identified (block 120). First, the components of block b_{ij} are passed through a noise shaping filter and scaled as set out in Equation 11 (block 121).

$$w_j = 0.875 * w_{j-1} - 0.5 * w_{j-2} + 0.4375 * w_{j-3} + b_{ij}, \qquad 0 \leq j < M$$

$$v_{ij} = G * w_j \qquad 0 \leq j < M$$

$$\text{Equation 11}$$

[0066] Thus, v_{ij} is the jth component of the vector v_i, and the values w_{-1}, w_{-2} and w_{-3} are the states of the noise shaping filter and are initialized to zero for each diphone. The filter coefficients are chosen to shape the quantization noise spectra in order to improve the subjective quality of the decompressed speech. After each vector is coded and decoded, these states are updated as described below with reference to blocks 124-126.

[0067] Next, the routine finds a pointer to the best match in a vector quantization table (block 122). The vector quantization table 123 consists of a sequence of vectors C_0 through C_{255} (block 123).

[0068] Thus, the vector v_i is compared against 256 M-point vectors, which are precomputed and stored in the code table 123. The vector C_{qi} which is closest to v_i is determined according to Equation 12. The value C_p for p = 0 through 255 represents the p^{th} encoding vector from the vector quantization code table 123.

$$\min_{p} \sum_{j=0}^{M-1} (v_{ij} - C_{pj})^2 \qquad \text{Equation 12}$$

[0069] The closest vector C_{qi} can also be determined efficiently using the technique of Equation 13.

$$v_i^T \bullet C_{qi} \leq v_i^T \bullet C_p \text{ for all } p (0 \leq p \leq 255) \qquad \text{Equation 13}$$

In Equation 13, the value v^T represents the transpose of the vector v, and "•" represents the inner product operation in the inequality.

[0070] The encoding vectors C_p in table 123 are utilized to match on the noise filtered value v_{ij}. However in decoding, a decoding vector table 125 is used which consists of a sequence of vectors QV_p. The values QV_p are selected for the purpose of achieving quality sound data using the vector quantization technique. Thus, after finding the vector C_{qi}, the pointer q is utilized to access the vector QV_{qi}. The decoded sample corresponding to the vector b_i which is produced at step 55 of Fig. 4, is the M-point vector (1/G) * QV_{qi}. The vector C_p is related to the vector QV_p by the noise shaping filter operation of Equation 11. Thus, when the decoding vector QV_p is accessed, no inverse noise shaping filter needs to be computed in the decode operation. The table 125 of Fig. 6 thus includes noise compensated quantization vectors.

[0071] In continuing to compute the encoding vectors for the vectors b_{ij}. which make up the residual signal r_n, the decoding vector of the pointer to the vector b_i is accessed (block 124). That decoding vector is used for filter and PBUF updates (block 126).

[0072] For the noise shaping filter, after the decoded samples are computed for each sub-block b_i, the error vector $(b_i\text{-}QV_{qi})$ is passed through the noise shaping filter as shown in Equation 14.

$$W_j = 0.875 * W_{j-1} - 0.5 * W_{j-2} + 0.4375 * W_{j-3} + [b_{ij} - QV_{qi}(j)] \qquad 0 \le j < M \qquad \text{Equation 14}$$

[0073] In Equation 14, the value $QV_{qi}(j)$ represents the j^{th} component of the decoding vector QV_{qi}. The noise shaping filter states for the next block are updated as shown in Equation 15.

$$w_{-1} = w_{M-1}$$

$$w_{-2} = w_{M-2}$$

$$w_{-3} = w_{M-3} \qquad \text{Equation 15}$$

[0074] This coding and decoding is performed for all of the N/M subblocks to obtain N/M indices to the decoding vector table 125. This string of indices Q_n, for n going from zero to N/M-1 represents identifiers for a string of decoding vectors for the residual signal r_n.

[0075] Thus, four parameters represent the N-point data sequence y_n:

1) Optimum pitch, P_{opt} (8 bits),
2) Pitch filter gain, β (4 bits),
3) Scaling parameter, G (3 bits), and
4) A string of decoding table indices, Q_n $(0 \le n < N/M)$.

[0076] The parameters β and G can be coded into a single byte. Thus, only (N/M) plus 2 bytes are used to represent N samples of speech. For example, suppose nominal pitch is 100 samples long, and M = 16. In this case, a frame of 96 samples of speech are represented by 8 bytes: 1 byte for P_{opt}, 1 byte for β and G, and 6 bytes for the decoding table indices Q_n. If the uncompressed speech consists of 16 bit samples, then this represents a compression of 24:1.

[0077] Back to Fig. 4, four parameters identifying the speech data are stored (block 57). In a preferred system, they are stored in a structure as described with respect to Fig. 3 where the structure of the frame can be characterized as follows:

```
#define      NumOfVectorsPerFrame   (FrameSize / VectorSize)

struct frame {
        unsigned     Gain : 4;
        unsigned     Beta : 3;
        unsigned     UnusedBit: 1;
        unsigned     char Pitch ;
        unsigned     char VQcodes[NumOfVectorsPerFrame]; };
```

[0078] The diphone record of Fig. 3 utilizing this frame structure can be characterized as follows:

```
DiphoneRecord
{
        char   LeftPhone, RightPhone;
        short  LeftPitchPeriodCount,RightPitchPeriodCount;
        short  *LeftPeriods, *RightPeriods;
        struct        frame *LeftData, *RightData;
}
```

[0079] These stored parameters uniquely provide for identification of the diphones required for text-to-speech synthesis.

[0080] As mentioned above with respect to Fig. 6, the encoder continues decoding the data being encoded in order to update the filter and PBUF values. The first step involved in this is an inverse pitch filter (block 58). With the vector r'_n corresponding to the decoded signal formed by concatenating the string of decoding vectors to represent the residual signal r'_n, the inverse filter is implemented as set out in Equation 16.

$$y'_n = r'_n + \beta * PBUF_{Pmax - Popt + n} \qquad 0 \le n < N. \qquad \text{Equation 16}$$

[0081] Next, the pitch buffer is updated (block 59) with the output of the inverse pitch filter. The pitch buffer PBUF is updated as set out in Equation 17.

$$PBUF_n = PBUF_{(n + N)} \qquad 0 \le n < (P_{max} - N)$$

$$PBUF_{(Pmax - N + n)} = y'_n \qquad 0 \le n < N \qquad \text{Equation 17}$$

[0082] Finally, the linear prediction filter parameters are updated using an inverse linear prediction filter step (block 60). The output of the inverse pitch filter is passed through a first order inverse linear prediction filter to obtain the decoded speech. The difference equation to implement this filter is set out in Equation 18.

$$x'_n = 0.875 * x'_{n-1} + y'_n \qquad \text{Equation 18}$$

[0083] In Equation 18, x'_n is the decompressed speech. From this, the value of x_{-1} for the next frame is set to the value x_N for use in the step of block 52.

[0084] Fig. 7 illustrates the decoder routine. The decoder module accepts as input (N/M) + 2 bytes of data, generated by the encoder module, and applies as output N samples of speech. The value of N depends on the nominal pitch of the speech data and the value of M depends on the desired compression ratio.

[0085] In software only text-to-speech systems, the computational complexity of the decoder must be as small as possible to ensure that the text-to-speech system can run in real time even on slow computers. A block diagram of the encoder is shown in Fig. 7.

[0086] The routine starts by accepting diphone records at block 200. The first step involves parsing the parameters G, β, P_{opt}, and the vector quantization string Q_n (block 201). Next, the residual signal r'_n is decoded (block 202). This involves accessing and concatenating the decoding vectors for the vector quantization string as shown schematically at block 203 with access to the decoding quantization vector table 125.

[0087] After the residual signal r'_n is decoded, an inverse pitch filter is applied (block 204). This inverse pitch filter is implemented as shown in Equation 19:

$$y'_n = r'_n + \beta * SPBUF(P_{max} - P_{opt} + n), 0 \le n < N. \qquad \text{Equation 19}$$

SPBUF is a synthesizer pitch buffer of length P_{max} initialized as zero for each diphone, as described above with respect to the encoder pitch buffer PBUF.

[0088] For each frame, the synthesis pitch buffer is updated (block 205). The manner in which it is updated is shown in Equation 20:

$$SPBUF_n = SPBUF_{(n+N)} \qquad 0 \le n < (P_{max} - N)$$

$$SPBUF_{(Pmax - N + n)} = y'_n \qquad 0 \le n < N \qquad \text{Equation 20}$$

[0089] After updating SPBUF, the sequence y'_n is applied to an inverse linear prediction filtering step (block 206). Thus, the output of the inverse pitch filter y'_n is passed through a first order inverse linear prediction filter to obtain the decoded speech. The difference equation to implement the inverse linear prediction filter is set out in Equation 21:

$$x'_n = 0.875 * x'_{n-1} + y'_n \qquad \text{Equation 21}$$

[0090] In Equation 21, the vector x'_n corresponds to the decompressed speech. This filtering operation can be implemented using simple shift operations without requiring any multiplication. Therefore, it executes very quickly and utilizes a very small amount of the host computer resources.

[0091] Encoding and decoding speech according to the algorithms described above, provide several advantages over prior art systems. First, this technique offers higher speech compression rates with decoders simple enough to be used in the implementation of software only text-to-speech systems on computer systems with low processing power. Second, the technique offers a very flexible trade-off between the compression ratio and synthesizer speech quality. A high-end computer system can opt for higher quality synthesized speech at the expense of a bigger RAM memory requirement.

III. <u>Waveform Blending For Discontinuity Smoothing</u> (Figs. 8 and 9)

[0092] As mentioned above with respect to Fig. 2, the synthesized frames of speech data generated using the vector quantization technique may result in slight discontinuities between diphones in a text string. Thus, the text-to-speech system provides a module for blending the diphone data frames to smooth such discontinuities. The blending technique of the preferred embodiment is shown with respect to Figs. 8 and 9.

[0093] Two concatenated diphones will have an ending frame and a beginning frame. The ending frame of the left diphone must be blended with the beginning frame of the right diphone without audible discontinuities or clicks being generated. Since the right boundary of the first diphone and the left boundary of the second diphone correspond to the same phoneme in most situations, they are expected to be similar looking at the point of concatenation. However, because the two diphone codings are extracted from different context, they will not look identical. This blending technique is applied to eliminate discontinuities at the point of concatenation. In Fig. 9, the last frame, referring here to one pitch period, of the left diphone is designated L_n ($0 \le n < PL$) at the top of the page. The first frame (pitch period) of the right diphone is designated R_n ($0 \le n < PR$). The blending of L_n and R_n according to the present invention will alter these two pitch periods only and is performed as discussed with reference to Fig. 8. The waveforms in Fig. 9 are chosen to illustrate the algorithm, and may not be representative of real speech data.

[0094] Thus, the algorithm as shown in Fig. 8 begins with receiving the left and right diphone in a sequence (block 300). Next, the last frame of the left diphone is stored in the buffer L_n (block 301). Also, the first frame of the right diphone is stored in buffer R_n (block 302).

[0095] Next, the algorithm replicates and concatenates the left frame L_n to form extend frame (block 303). In the next step, the discontinuities in the extended frame between the replicated left frames are smoothed (block 304). This smoothed and extended left frame is referred to as El_n in Fig. 9.

[0096] The extended sequence El_n ($0 \le n < PL$) is obtained in the first step as shown in Equation 22:

$$El_n = L_n \qquad n = 0,1,...,PL-1$$

$$El_{PL+n} = L_n \qquad n = 0,1,...,PL-1 \qquad \text{Equation 22}$$

Then discontinuity smoothing from the point $n = P^L$ is conducted according to the filter of Equation 23:

$$El_{PL+n} = El_{PL+n} + [El_{(PL-1)} - El'_{(PL-1)}]^{*}\Delta^{n+1}, \qquad n = 0,1,...,(PL/2). \qquad \text{Equation 23}$$

In Equation 23, the value Δ is equal to 15/16 and $El'_{(PL-1)} = El_2 + 3 * (El_1 - El_0)$. Thus, as indicated in Fig. 9, the extended sequence El_n is substantially equal to L_n on the left hand side, has a smoothed region beginning at the point P_L and converges on the original shape of L_n toward the point $2P_L$. If L_n was perfectly periodic, then $El_{PL-1} = El'_{PL-1}$.

[0097] In the next step, the optimum match of R_n with the vector El_n is found. This match point is referred to as P_{opt}. (Block 305.) This is accomplished essentially as shown in Fig. 9 by comparing R_n with El_n to find the section of El_n which most closely matches R_n. This optimum blend point determination is performed using Equation 23 where W is the minimum of PL and PR, and AMDF represents the average magnitude difference function.

$$\mathrm{AMDF}(p) = \sum_{n=0}^{W-1} \left| El_{n+p} - R_n \right| \qquad \text{Equation 24}$$

[0098] This function is computed for values of p in the range of 0 to PL-1. The vertical bars in the operation denote the absolute value. W is the window size for the AMDF computation. P_{opt} is chosen to be the value at which AMDF(p) is minimum. This means that $p = P_{opt}$ corresponds to the point at which sequences $El_{n+p}(0 \le n < W)$ and $R_n(0 \le n < W)$ are very close to each other.

[0099] After determining the optimum blend point P_{opt}, the waveforms are blended (block 306). The blending utilizes a first weighting ramp WL which is shown in Fig. 9 beginning at P_{opt} in the El_n trace. In a second ramp, WR is shown in Fig. 9 at the R_n trace which is lined up with P_{opt}. Thus, in the beginning of the blending operation, the value of El_n is emphasized. At the end of the blending operation, the value of R_n is emphasized.

[0100] Before blending, the length PL of L_n is altered as needed to ensure that when the modified L_n and R_n are concatenated, the waveforms are as continuous as possible. Thus, the length P'L is set to P_{opt} if P_{opt} is greater than PL/2. Otherwise, the length P'L is equal to W + P_{opt} and the sequence L_n is equal to El_n for $0 \le n \le (P'L-1)$.

[0101] The blending ramp beginning at P_{opt} is set out in Equation 25:

$$R_n = El_{n+Popt} + (R_n - El_{n+Popt})*(n+1)/W \qquad 0 \le n < W$$

$$R_n = R_n \qquad W \le n < PR \qquad \text{Equation 25}$$

[0102] Thus, the sequences L_n and R_n are windowed and added to get the blended R_n. The beginning of L_n and the ending of R_n are preserved to prevent any discontinuities with adjacent frames.

[0103] This blending technique is believed to minimize blending noise in synthesized speech produced by any concatenated speech synthesis.

IV. Pitch and Duration Modification (Figs. 10-18)

[0104] As mentioned above with respect to Fig. 2, a text analysis program analyzes the text and determines the duration and pitch contour of each phone that needs to be synthesized and generates intonation control signals. A typical control for a phone will indicate that a given phoneme, such as AE, should have a duration of 200 milliseconds and a pitch should rise linearly from 220Hz to 300Hz. This requirement is graphically shown in Fig. 10. As shown in Fig. 10, T equals the desired duration (e.g. 200 milliseconds) of the phoneme. The frequency f_b is the desired beginning pitch in Hz. The frequency f_e is the desired ending pitch in Hz. The labels P_1, P_2..., P_6 indicate the number of samples of each frame to achieve the desired pitch frequencies f_b, f_2...,f_6. The relationship between the desired number of samples, P_i, and the desired pitch frequency f_i ($f_1 = f_b$), is defined by the relation:

$P_i = F_s/f_i$, where F_s is the sampling frequency for the data.

As can be seen in Fig. 10, the pitch period for a lower frequency period of the phoneme is longer than the pitch period for a higher frequency period of the phoneme. If the nominal frequency were P_3, then the algorithm would be required to lengthen the pitch period for frames P_1 and P_2 and decrease the pitch periods for frames P_4, P_5 and P_6. Also, the given duration T of the phoneme will indicate how many pitch periods should be inserted or deleted from the encoded phoneme to achieve the desired duration period. Figs. 11 through 18 illustrate a preferred implementation of such algorithms.

[0105] Fig. 11 illustrates an algorithm for increasing the pitch period, with reference to the graphs of Fig. 12. The algorithm begins by receiving a control to increase the pitch period to N + Δ, where N is the pitch period of the encoded frame. (Block 350). In the next step, the pitch period data is stored in a buffer x_n (block 351). x_n is shown in Fig. 12 at the top of the page. In the next step, a left vector L_n is generated by applying a weighting function WL to the pitch period data x_n with reference to Δ (block 352). This weighting function is illustrated in Equation 26 where M = N-Δ:

$$L_n = x_n \qquad \text{for } 0 \leq n < \Delta$$

$$L_n = x_n * (N-n)/(M+1) \qquad \text{for } \Delta \leq n < N \qquad \text{Equation 26}$$

As can be seen in Fig. 12, the weighting function WL is constant from the first sample to sample Δ, and decreases from Δ to N.

[0106] Next, a weighting function WR is applied to x_n (block 353) as can be seen in the Fig. 12. This weighting function is executed as shown in Equation 27:

$$R_n = x_{n+\Delta} *(n+1)/(M+1) \qquad \text{for } 0 \leq n < N-\Delta$$

$$R_n = x_{n+\Delta} \qquad \text{for } N-\Delta \leq n < N. \qquad \text{Equation 27}$$

[0107] As can be seen in Fig. 12, the weighting function WR increases from 0 to N-Δ and remains constant from N-Δ to N. The resulting waveforms L_n and R_n are shown conceptually in Fig. 12. As can be seen, L_n maintains the beginning of the sequence x_n, while R_n maintains the ending of the data x_n.

[0108] The pitch modified sequence y_n is formed (block 354) by adding the two sequences as shown in Equation 28:

$$y_n = L_n + R_{(n-\Delta)} \qquad \text{Equation 28}$$

This is graphically shown in Fig. 12 by placing R_n shifted by Δ below L_n. The combination of L_n and R_n shifted by Δ is shown to be y_n at the bottom of Fig. 12. The pitch period for y_n is N + Δ. The beginning of y_n is the same as the beginning of x_n, and the ending of y_n is substantially the same as the ending of x_n. This maintains continuity with adjacent frames in the sequence, and accomplishes a smooth transition while extending the pitch period of the data.

[0109] Equation 28 is executed with the assumption that L_n is 0, for $n \leq N$, and R_n is 0 for n<0. This is illustrated pictorially in Fig. 12.

[0110] An efficient implementation of this scheme which requires at most one multiply per sample, is shown in Equation 29:

$$y_n = x_n \qquad 0 \leq n < \Delta$$

$$y_n = x_n + [x_{n-\Delta} - x_n]*(n-\Delta + 1)/(N-\Delta + 1) \qquad \Delta \leq n < N$$

$$y_n = x_{n-\Delta} \qquad N \leq n < N_d \qquad \text{Equation 29}$$

This results in a new pitch period having a pitch period of N + Δ.

[0111] There are also instances in which the pitch period must be decreased. The algorithm for decreasing the pitch period is shown in Fig. 13 with reference to the graphs of Fig. 14. Thus, the algorithm begins with a control signal indicating that the pitch period must be decreased to N-Δ. (Block 400). The first step is to store two consecutive pitch periods in the buffer x_n (block 401). Thus, the buffer x_n as can be seen in Fig. 14 consists of two consecutive pitch periods, with the period N_l being the length of the first pitch period, and N_r being the length of the second pitch period. Next, two sequences L_n and R_n are conceptually created using weighting functions WL and WR (blocks 402 and 403). The weighting function WL emphasizes the beginning of the first pitch period, and the weighting function WR emphasizes the ending of the second pitch period. These functions can be conceptually represented as shown in Equations 30 and 31, respectively:

$$L_n = x_n \qquad \text{for } 0 \leq n < N_l - W$$

$$L_n = x_n * (N_l\text{-}n)/(W+1) \qquad W \le n < N_l$$

$$L_n = 0 \qquad \text{otherwise.} \qquad \text{Equation 30}$$

and

$$R_n = x_n * (n\text{-}N_l\text{+}W\text{-}\Delta + 1)/(W + 1) \qquad \text{for } N_l\text{-}W + \Delta \le n < N_l + \Delta$$

$$R_n = x_n \qquad \text{for } N_l + \Delta \le n < N_l + N_r$$

$$R_n = 0 \qquad \text{otherwise.} \qquad \text{Equation 31}$$

[0112] In these equations, Δ is equal to the difference between N_l and the desired pitch period N_d. The value W is equal to 2*Δ, unless 2*Δ is greater than N_d, in which case W is equal to N_d.

[0113] These two sequences L_n and R_n are blended to form a pitch modified sequence y_n (block 404). The length of the pitch modified sequence y_n will be equal to the sum of the desired length and the length of the right phoneme frame N_r. It is formed by adding the two sequences as shown in Equation 32:

$$y_n = L_n + R_{(n + \Delta)} \qquad \text{Equation 32}$$

[0114] Thus, when a pitch period is decreased, two consecutive pitch periods of data are affected, even though only the length of one pitch period is changed. This is done because pitch periods are divided at places where short-term energy is the lowest within a pitch period. Thus, this strategy affects only the low energy portion of the pitch periods. This minimizes the degradation in speech quality due to the pitch modification. It should be appreciated that the drawings in Fig. 14 are simplified and do not represent actual pitch period data.

[0115] An efficient implementation of this scheme, which requires at most one multiply per sample, is set out in Equations 33 and 34.

[0116] The first pitch period of length N_d is given by Equation 33:

$$y_n = x_n \qquad 0 \le n < N_l\text{-}W$$

$$y_n = x_n + [x_{n+\Delta} - x_n]*(n\text{-}N_l + W + 1)/(W + 1) \qquad N_l\text{-}W \le n < N_d \qquad \text{Equation 33}$$

[0117] The second pitch period of length N_r is generated as shown in Equation 34:

$$y_n = x_{n\text{-}\Delta} + [x_n - x_{n\text{-}\Delta}]*(n\text{-}\Delta\text{-}N_l + W + 1)/(W + 1) \qquad N_l \le n < N_l + \Delta$$

$$y_n = x_n \qquad N_{l+\Delta} \le n < N_l + N_r$$

$$\text{Equation 34}$$

[0118] As can be seen in Fig. 14, the sequence L_n is essentially equal to the first pitch period until the point N_l-W. At that point, a decreasing ramp WL is applied to the signal to dampen the effect of the first pitch period.

[0119] As also can be seen, the weighting function WR begins at the point N_l-W + Δ and applies an increasing ramp to the sequence x_n until the point N_l + Δ. From that point, a constant value is applied. This has the effect of damping the effect of the right sequence and emphasizing the left during the beginning of the weighting functions, and generating an ending segment which is substantially equal to the ending segment of x_n emphasizing the right sequence and damping the left. When the two functions are blended, the resulting waveform y_n is substantially equal to the beginning of x_n at the beginning of the sequence, at the point N_l-W a modified sequence is generated until the point N_l. From N_l to the ending, sequence x_n is shifted by Δ results.

[0120] A need also arises for insertion of pitch periods to increase the duration of a given sound. A pitch period is inserted according to the algorithm shown in Fig. 15 with reference to the drawings of Fig. 16.

[0121] The algorithm begins by receiving a control signal to insert a pitch period between frames L_n and R_n (block 450). Next, both L_n and R_n are stored in the buffer (block 451), where L_n and R_n ar two adjacent pitch periods of a voice diphone. (Without loss of generality, it is assumed for the description that the two sequences are of equal lengths N.)

[0122] In order to insert a pitch period, x_n of the same duration, without causing a discontinuity between L_n and x_n and between x_n and R_n, the pitch period x_n should resemble R_n around n = 0 (preserving L_n to x_n continuity), and should resemble L_n around n=N (preserving x_n to R_n continuity). This is accomplished by defining x_n as shown in Equation 35:

$$x_n = R_n + (L_n - R_n) * [(n + 1)/(N + 1)] \qquad 0 \leq n < N-1 \qquad \text{Equation 35}$$

[0123] Conceptually, as shown in Fig. 15, the algorithm proceeds by generating a left vector WL(L_n), essentially applying to the increasing ramp WL to the signal L_n. (Block 452).

[0124] A right vector WR (R_n) is generated using the weighting vector WR (block 453) which is essentially a decreasing ramp as shown in Fig. 16. Thus, the ending of L_n is emphasized with the left vector, and the beginning of R_n is emphasized with the vector WR.

[0125] Next, WR (L_n) and WR (R_n) are blended to create an inserted period x_n (block 454).

[0126] The computation requirement for inserting a pitch period is thus just a multiplication and two additions per speech sample.

[0127] Finally, concatenation of L_n, x_n and R_n produces a sequence with an inserted pitch period (block 455).

[0128] Deletion of a pitch period is accomplished as shown in Fig. 17 with reference to the graphs of Fig. 18. This algorithm, which is very similar to the algorithm for inserting a pitch period, begins with receiving a control signal indicating deletion of pitch period R_n which follows L_n (block 500). Next, the pitch periods L_n and R_n are stored in the buffer (block 501). This is pictorially illustrated in Fig. 18 at the top of the page. Again, without loss of generality, it is assumed that the two sequences have equal lengths N.

[0129] The algorithm operates to modify the pitch period L_n which precedes R_n (to be deleted) so that it resembles R_n, as n approaches N. This is done as set forth in Equation 36:

$$L'_n = L_n + (R_n - L_n) * [(n + 1)/(N + 1)] \qquad 0 \leq n < N-1 \qquad \text{Equation 36}$$

In Equation 36, the resulting sequence L'_n is shown at the bottom of Fig. 18. Conceptually, Equation 36 applies a weighting function WL to the sequence L_n (block 502). This emphasizes the beginning of the sequence L_n as shown. Next, a right vector WR (R_n) is generated by applying a weighting vector WR to the sequence R_n that emphasizes the ending of R_n (block 503).

[0130] WL (L_n) and WR (R_n) are blended to create the resulting vector L'_n. (Block 504). Finally, the sequence L_n-R_n is replaced with the sequence L'_n in the pitch period string. (Block 505).

IV. Conclusion

[0131] Accordingly, the present invention presents a software only text-to-speech system which is efficient, uses a very small amount of memory, and is portable to a wide variety of standard microcomputer platforms. It takes advantage of knowledge about speech data, to create a speech compression, blending, and duration control routine which produces very high quality speech with very little computational resources.

[0132] A source code listing of the software for executing the compression and decompression, the blending, and the duration and pitch control routines is provided in the Appendix as an example of a preferred embodiment of the present invention.

[0133] The foregoing description of preferred embodiments of the present invention has been provided for the purposes of illustration and description. It is not intended to be exhaustive or to limit the invention to the precise forms disclosed. Obviously, many modifications and variations will be apparent to practitioners skilled in this art. The embodiments were chosen and described in order to best explain the principles of the invention and its practical application, thereby enabling others skilled in the art to understand the invention for various embodiments and with various modifications as are suited to the particular use contemplated. It is intended that the scope of the invention be defined by the following claims.

APPENDIX

COMPUTER PROGRAM LISTINGS

TABLE OF CONTENTS

I. ENCODER MODULE

```
#include <stdio.h>
#include <math.h>
#include <StdLib.h>
#include <types.h>
#include <fcntl.h>
#include <string.h>

#include <types.h>
#include <files.h>
#include <resources.h>
#include <memory.h>
#include "vqcoder.h"

#define     LAST_FRAME_FLAG          128
#define     PBUF_SIZE  440
static  float          oc_state[2], nsf_state[NSF_ORDER+1];
static  short          pstate[PORDER+1], dstate[PORDER+1];
static  short          AnaPbuf[PBUF_SIZE];

static  short          vsize, cbook_size, bs_size;

#pragma segment vqlib

/* Read Code Books */
float   *EncodeBook[MAX_CBOOK_SIZE];
short   *DecodeBook[MAX_CBOOK_SIZE];
get_cbook(short ratio)
{
   short *p;
   short    frame_size, i;
   static   short last_ratio = 0;

   Handle h;
   int      skip;
   h = GetResource('CBOK',1);
   HLock(h);
   p = (short *) *h;

   if (ratio = = last_ratio)
       return;
   last_ratio = ratio;

   if (ratio < 3)
       return;

   if (NOMINAL_PITCH < 165)
```

```
        frame_size = 96;
    else
        frame_size = 160;

    get_compr_pars(ratio, frame_size, &vsize, &cbook_size, &bs_size);
    skip = 0;
    while (p[skip + 1] ! = vsize)
    {
        short t1, t2;
        t2 = p[skip];
        t1 = p[skip + 1];
        skip + = sizeof(float) * (2 * t2-1) * (t1 + 1) / sizeof(short)
            + (2 * t2 * t1 + 2);
    }

    /*Skip Binary search tree */
    skip + = sizeof(float) * (cbook_size-1) * (vsize + 1) / sizeof(short)
        + (cbook_size * vsize + 2);

    /* Get pointers to Full search code books */
    for (i = 0; i < cbook_size; i+ +)
    {
        EncodeBook[i] = (float *) &p[skip];
        skip + = (vsize + 1) * sizeof (float) / sizeof(short);
    }

    for (i = 0; i < cbook_size; i+ +)
    {
        DecodeBook[i] = p + skip;
        skip + = vsize;
    }
}

char *getcbook(long *len, short ratio)
{
    get_cbook(ratio);
    *len = sizeof(short) * vsize * cbook_size;
        /* plus one is to make space at the end for the array of pointers */
    return (char*) DecodeBook[0];
}

/* A Routine for Pitch filter parameter Estimation */
GetPitchFilterPars (x, len, pbuf, min_pitch, max_pitch, pitch, beta)
float   *beta;
short   *x, *pbuf;
short   min_pitch, max_pitch;
short   len;
unsigned int *pitch;
{
    /* Estimate long-term predictor */
    int     best_pitch, i, j;
    float   syy, sxy, best_sxy = 0.0, best_syy = 1.0;
    short   *ptr;

    best_pitch = min_pitch;
    ptr = pbuf + PBUF_SIZE - min_pitch;
    syy = 1.0;
    for (i = 0; i < len; i++)
    {
            syy += (*ptr) * (*ptr);
            ptr++;
    }
    for (j = min_pitch; j < max_pitch; j++)
    {
        sxy = 0.0;
        ptr = pbuf + PBUF_SIZE - j;
        for (i = 0; i < len; i++)
            sxy += x[i] * (*ptr++);

        if (sxy > 0 && (sxy * sxy * best_syy > best_sxy * best_sxy * syy))
        {
            best_syy = syy;
            best_sxy = sxy;
            best_pitch = j;
        }
        syy = syy - pbuf[PBUF_SIZE - j + len - 1] * pbuf[PBUF_SIZE - j + len - 1]
                + pbuf[PBUF_SIZE - j - 1] * pbuf[PBUF_SIZE - j - 1];
    }

    *pitch = best_pitch;
    *beta = best_sxy / best_syy;
}

/* Quantization of LTP gain parameter */
CodePitchFilterGain(beta, bcode)
float beta;
unsigned int *bcode;
{
    int i;
    for (i = 0; i < DLB_TAB_SIZE; i++)
    {
       if (beta <= dlb_tab[i])
          break;
    }
    *bcode = i;
}

/* Pitch filter */
PitchFilter(data, len, pbuf, pitch, ibeta)
float   *data;
```

```
short   ibeta;
short   *pbuf;
short       len;
unsigned int pitch;
{

  long          pn;
  int        i, j;


   j = PBUF_SIZE - pitch;
   for (i = 0; i < len; i++)
   {
       pn  = ((ibeta * pbuf[j++]) >> 4);
       data[i] -= pn;
   }
}

/* Forward Noise Shaping filter */
FNSFilter(float *inp, float *state, short len, float *out)
{
   short i, j;
   for (j = 0; j < len; j++)
   {
       float tmp = inp[j];
       for (i = 1; i <= NSF_ORDER; i++)
           tmp += state[i] * nsf[i];
       out[j] = state[0] = tmp;
       for (i = NSF_ORDER; i > 0; i--)
           state[i] = state[i-1];
   }
}

/* Update Noise shaping Filter states */
UpdateNSFState(float *inp, float *state, short len)
{
   short i, j;
   float     temp_state[NSF_ORDER + 1];

   for (i = 0; i <= NSF_ORDER; i++)
       temp_state[i] = 0;

   for (j = 0; j < len; j++)
   {
       float tmp = inp[j];
       for (i = 1; i <= NSF_ORDER; i++)
           tmp += temp_state[i] * nsf[i];
       temp_state[0] = tmp;
       for (i = NSF_ORDER; i > 0; i--)
           temp_state[i] = temp_state[i-1];
   }
   for (i = 0; i < = NSF_ORDER; i+ + ).
       state[i] = state[i] - temp_state[i];
}

/* Quantization of Segment Power */
CodeBlockGain(power, gcode)
float power;
unsigned int *gcode;
{
   int i;
   for (i = 0; i < DLG_TAB_SIZE; i+ +)
   {
      if (power < = dlg_tab[i])
         break;
   }
   *gcode = i;
}

/* Full search Coder */
VQCoder(float *x, float *nsf_state, short len, struct frame *bs)
{
   float                 max_x, tmp;
   int              i, j, k, index, lshift_count;
   unsigned int       gcode;
   float                 min_err = 0;

   max_x = x[0];
   for (i = 1; i < len; i+ +)
       if ( fabs(x[i]) > max_x)
           max_x = fabs(x[i]);

   CodeBlockGain(max_x, &gcode);
   max_x = qlg_tab[gcode];
   lshift_count = 7 - gcode;           /* To scale 14-bit Code book output to the 16-bit
actual value */
   bs->gcode = gcode;

   for (i = 0; i < len; i + = vsize)
   {
       /* Filter the data vector */
       FNSFilter(&x[i], nsf_state, vsize, &x[i]);

       /* Scale data */
       for (j = i; j < i + vsize; j+ +)
           x[j] = x[j] * 1024 / max_x;

       index = 0;
       for (j = 0; j < cbook_size; j+ +)
       {
            tmp = EncodeBook[j][vsize] * 1024.0;
            for (k = 0; k < vsize; k++)
                tmp -= x[i+k] * EncodeBook[j][k];

            if (tmp < min_err || j == 0)
            {
                index = j;
                min_err = tmp;
            }
        }
        bs->vqcode[i/vsize] = index;

        /* Rescale data: Decoded data is 14-bits, convert to 16 bits */
        if (lshift_count)
        {
            for (k = 0; k < vsize; k++)
                x[i+k] = ((4 * DecodeBook[index][k]) >> lshift_count);
        }
        else
        {
            for (k = 0; k < vsize; k++)
                x[i+k] = 4 * DecodeBook[index][k];
        }

        /* Update noise shaping filter state */
        UpdateNSFState(&x[i], nsf_state, vsize);
    }
}

init_compress()
{
    int i;
    oc_state[0] = 0;;
    oc_state[1] = 0;;
    for (i = 0; i <= PORDER; i++)
        pstate[i] = dstate[i] = 0;
    for (i = 0; i < PBUF_SIZE; i++)
        AnaPbuf[i] = 0;
    for (i=0; i <= NSF_ORDER; i++)
        nsf_state[i] = 0;
}

Encoder(xn, frame_size, min_pitch, max_pitch, bs)
short xn[];
struct frame *bs;
short  frame_size, min_pitch, max_pitch;
{
    unsigned int pitch, bcode;
    float     preemp_xn[PBUF_SIZE], beta;
    short     xn_copy[PBUF_SIZE];
short    ibeta;
float    acc;
int i, j;


/* Offset Compensation */
for (i = 0; i < frame_size; i+ +)
{
    float inp = xn[i];
    xn[i] = inp - oc_state[0] + ALPHA * oc_state[1];
    oc_state[1] = xn[i];
    oc_state[0] = inp;
}

/* Linear Prediction Filtering */
for (i = 0; i < frame_size; i+ +)
{
    acc = pstate[0] = xn[i];
    for (j = 1; j < = PORDER; j+ +)
        acc - = pstate[j] * pfilt[j];
    xn_copy[i] = preemp_xn[i] = acc;
    for (j = PORDER; j > 0; j--)
        pstate[j] = pstate[j-1];
}

GetPitchFilterPars (xn_copy, frame_size, AnaPbuf, min_pitch,
    max_pitch, &pitch, &beta);
CodePitchFilterGain(beta, &bcode);
ibeta = qlb_tab[bcode];

bs->bcode = bcode;
bs->pitch = pitch - min_pitch + 1;

PitchFilter(preemp_xn, frame_size, AnaPbuf, pitch, ibeta);

VQCoder(preemp_xn, nsf_state, frame_size, bs);

/* Inverse Filtering */
j = PBUF_SIZE - pitch;
for (i = 0; i < frame_size; i+ +)
{
    xn_copy[i] = preemp_xn[i];
    xn_copy[i] + = ((ibeta * AnaPbuf[j+ +]) > > 4);
}

/* Update Pitch Buffer */
j = 0;
for (i = frame_size; i < PBUF_SIZE; i+ +)
    AnaPbuf[j+ +] = AnaPbuf[i];
for (i = 0; i < frame_size; i+ +)
```

```
            AnaPbuf[j++] = xn_copy[i];

        /* Inverse LP filtering  */
        for (i = 0; i < frame_size; i++)
        {
            acc = xn_copy[i];
            for (j = 1; j <= PORDER; j++)
                acc = acc + dstate[j] * pfilt[j];
            dstate[0] = acc;
            for (j = PORDER; j > 0; j--)
                dstate[j] = dstate[j-1];
        }

        for (j = 0; j <= PORDER; j++)
            pstate[j] = dstate[j];
    }

    compress (short *input, short ilen, unsigned char *output, long *olen, long docomp)
    {
        int             i, j, vcount;
        unsigned char   temp;
        short           frame_size, min_pitch, max_pitch;

        if (docomp > 2)
        {
            init_compress();

            if (NOMINAL_PITCH < 165)
            {
                min_pitch = 96;
                frame_size = 96;
                max_pitch = 350;
            }
            else
            {
                min_pitch = 160;
                frame_size = 160;
                max_pitch = 414;
            }

            bs_size = frame_size / vsize + 2;
            /* TEMPORARY: Storing State information */
            pstate[1] = *(input - 1);
            if (pstate[1] > 0)
                pstate[1] = (pstate[1] + 128) / 256 + 128;
            else
                pstate[1] = (pstate[1] - 128) / 256 + 128;

            if (pstate[1] < 0)
                pstate[1] = 0;
        if (pstate[1] > 255)
            pstate[1] = 255;
        *output = pstate[1];
        j = 1;
        pstate[1] = pstate[1] - 128;
        pstate[1] = 256 * pstate[1];
        dstate[1] = pstate[1];
        /* End of Hack */
        for (i = 0; i < ilen; i + = frame_size)
        {
            Encoder(input+i, frame_size, min_pitch, max_pitch, output+j);
            j + = bs_size;
        }
        j -= bs_size;

        /* Number of vectors in last frame */
        vcount = (ilen + frame_size - i + vsize - 1) / vsize;
        temp = output[j];
        output[j] = vcount + LAST_FRAME_FLAG;
        output[j + vcount + 2] = temp;
        *olen = j + vcount + 3;
    }
    else
    {
        static long SampCount = 0;
        copy(input, output, 2*ilen);
        SampCount + = ilen;
        *olen = ilen;
    }
}

copy(a, b, len)
short  *a, *b;
short  len;
{
    int i;
    for (i = 0; i < len; i++)
        *b++ = (*a++);
}
```

II. DECODER MODULE

```
#include <Types.h>
#include <Memory.h>
#include <Quickdraw.h>
#include <ToolUtils.h>
#include <errors.h>
#include <files.h>

#include "vtcint.h"
#include <stdlib.h>
#include <math.h>
#include <sysequ.h>
#include <string.h>

#define MAX_CBOOK_SIZE              256
#define    LAST_FRAME_FLAG          128
#define    PORDER                      1
#define    IPCONS                      7                     /* 7/8 */

#define    LARGE_NUM                   100000000
#define    VOICED      1

#define LEFT                    0
#define    RIGHT                    1
#define    UNVOICED                 0

#define    PFILT_ORDER                 8

struct frame {
 unsigned  gcode : 4;
 unsigned      bcode : 4;
 unsigned pitch : 8;
 unsigned char vqcode[];
};

void expand(short **DecodeBook, short frame_size, short vsize,
   short min_pitch, struct frame *bs, short *output, short smpnum);

get_compr_pars(short ratio, short frame_size, short *vsize,
   short *cbook_size, short *bs_size)
{
   switch (ratio)
   {
        case 4:
             *vsize = 2;
             *cbook_size = 256;
             *bs_size = frame_size/2 + 2;
             break;
        case 7:
            *vsize = 4;
            *cbook_size = 256;
            *bs_size = frame_size/4 + 2;
            break;
        case 14:
            *vsize = 8;
            *cbook_size = 256;
            *bs_size = frame_size/8 + 2;
            break;
        case 24:
            *vsize = 16;
            *cbook_size = 256;
            *bs_size = frame_size/16 + 2;
            break;
        default:
            *vsize = 2;
            *cbook_size = 256;
            *bs_size = frame_size/2 + 2;
            break;
    }
}

short *SnInit(short comp_ratio)
{
    short *state, *ptr;
    int i;

    state = ptr = (short*)NewPtr((PFILT_ORDER+1 + PFILT_ORDER/2 + 2) *
sizeof(short));
    if ( state == nil )
    {
        return nil;
    }
    for (i=0;i<PFILT_ORDER+1;i++)
        *ptr++ = 0;
/*
    if (comp_ratio == 24)
    {
        *ptr++ = 0.036953 * 32768 + 0.5;
        *ptr++ = -0.132232 * 32768 - 0.5;
        *ptr++ = 0.047798 * 32768 + 0.5;
        *ptr++ = 0.403220 * 32768 + 0.5;
        *ptr++ = 0.290033 * 32768 + 0.5;
    }
    else
    {
        *ptr++ = 0.074539 * 32768 + 0.5;
        *ptr++ = -0.174290 * 32768 - 0.5;
        *ptr++ = 0.013704 * 32768 + 0.5;
```

```
            *ptr+ + = 0.426815 * 32768 + 0.5;
            *ptr+ + = 0.320707 * 32768 + 0.5;
        }
    */
        if (comp_ratio = = 24)
        {
            *ptr+ + = 1211;
            *ptr+ + = -4333;
            *ptr+ + = 1566;
            *ptr+ + = 13213;
            *ptr+ + = 9504;
        }
        else
        {
            *ptr+ + = 2442;
            *ptr+ + = -5711;
            *ptr+ + = 449;
            *ptr+ + = 13986;
            *ptr+ + = 10509;
        }
        *ptr = 0;        /* DC value */
        return state;
    }

    SnDone(char *state)
    {
        if ( state ! = nil )
        {
            DisposPtr(state);
        }
    }

    short **SnDeInit(p, ratio,  frame_size)
    short *p,ratio, frame_size;
    {
        int i;
        short cbook_size = 256, vsize = 16, bs_size;
        short **DecodeBook;

        get_compr_pars(ratio, frame_size, &vsize, &cbook_size, &bs_size);

        DecodeBook = (short**)NewPtr(cbook_size * sizeof(short*));
        if (DecodeBook) {
            for (i = 0; i < cbook_size; i+ +)
            {
                DecodeBook[i] = p;
                p + = vsize;
            }
        }
        return DecodeBook;
```

```
}

SnDeDone(char *DecodeBook)
{
    if ( DecodeBook != nil )
    {
        DisposPtr(DecodeBook);
    }
}

void
expand(short **DecodeBook, short frame_size, short vsize,
    short min_pitch, struct frame *bs, short *output, short smpnum)
{
    short     count;
    short     *bptr, *sptr1, *sptr2;
    unsigned short pitch, bcode;
/*
    short qlb_tab[] = {
    1, 2, 3, 4, 5, 6, 7, 8,
    9, 10, 11, 12, 13, 14, 15, 16
    };
*/
    bcode = bs->bcode;
    pitch = bs->pitch + min_pitch - 1;

    /* Decode VQ vectors */
    {
        unsigned     char     *cptr;
        short     k, vsize_by_2;
        short     rshift_count = 7 - bs->gcode;    /* We want the output to be 14-bit
number */

        sptr1 = output + smpnum;
        cptr = bs->vqcode;
        vsize_by_2 = (vsize >> 1) + 1; /* +1 since we do a while (--i) instead of
while (i--) */
        if (rshift_count)
        {
            for (k = 0; k < frame_size; k += vsize)
            {
                bptr = DecodeBook[*cptr++];
                count = vsize_by_2;
                while (--count)
                {
                    *sptr1++ = ((*bptr++) >> rshift_count);
                    *sptr1++ = ((*bptr++) >> rshift_count);
                }
            }
        }
        else
        {
            for (k = 0; k < frame_size; k += vsize)
            {
                bptr = DecodeBook[*cptr++];
                count = vsize_by_2;
                while (--count)
                {
                    *sptr1++ = *bptr++;
                    *sptr1++ = *bptr++;
                }
            }
        }
    }

    /* Inverse Filtering */
    if (smpnum < pitch)
    {
        sptr1 = output + pitch;
        count = smpnum + frame_size + 1 - pitch; /* +1 since we do a while (--i)
 instead of while (i--) */
        sptr2 = sptr1 - pitch;
        switch (bcode)
        {
            case 0:
                while (--count)
                    *sptr1++ += ((*sptr2++) >> 4);
                break;
            case 1:
                while (--count)
                    *sptr1++ += ((*sptr2++) >> 3);
                break;
            case 2:
                while (--count)
                    *sptr1++ += ((3 * (*sptr2++)) >> 4);
                break;
            case 3:
                while (--count)
                    *sptr1++ += ((*sptr2++) >> 2);
                break;
            case 4:
                while (--count)
                    *sptr1++ += ((5 * (*sptr2++)) >> 4);
                break;
            case 5:
                while (--count)
                    *sptr1++ += ((3 * (*sptr2++)) >> 3);
                break;
            case 6:
                while (--count)
            *sptr1++ += ((7 * (*sptr2++)) >> 4);
        break;
    case 7:
        while (--count)
            *sptr1++ += ((*sptr2++) >> 1);
        break;
    case 8:
        while (--count)
        {
            long    tmp;
            tmp = *sptr2++;
            *sptr1++ += (((tmp << 3) + tmp) >> 4);
        }
        break;
    case 9:
        while (--count)
            *sptr1++ += ((5 * (*sptr2++)) >> 3);
        break;
    case 10:
        while (--count)
        {
            long    tmp;
            tmp = *sptr2++;
            *sptr1++ += (((tmp << 3) + 3 * tmp) >> 4);
        }
        break;
    case 11:
        while (--count)
            *sptr1++ += ((3 * (*sptr2++)) >> 2);
        break;
    case 12:
        while (--count)
        {
            long    tmp;
            tmp = *sptr2++;
            *sptr1++ += (((tmp << 4) - 3 * tmp) >> 4);
        }
        break;
    case 13:
        while (--count)
            *sptr1++ += ((7 * (*sptr2++)) >> 3);
        break;
    case 14:
        while (--count)
        {
            long    tmp;
            tmp = *sptr2++;
            *sptr1++ += (((tmp << 4) - tmp) >> 4);
        }
        break;
        case 15:
            while.(--count)
                *sptr1 + +  + = *sptr2 + +;
            break;
    }
} else {
    sptr1 = output + smpnum;
    sptr2 = sptr1 - pitch;
    count = (frame_size / 4) + 1;
    switch (bcode)
    {
        case 0:
            while (--count) {
                *sptr1 + +  + = ((*sptr2 + +) > > 4);
                *sptr1 + +  + = ((*sptr2 + +) > > 4);
                *sptr1 + +  + = ((*sptr2 + +) > > 4);
                *sptr1 + +  + = ((*sptr2 + +) > > 4);
            }
            break;
        case 1:
            while (--count) {
                *sptr1 + +  + = ((*sptr2 + +) > > 3);
                *sptr1 + +  + = ((*sptr2 + +) > > 3);
                *sptr1 + +  + = ((*sptr2 + +) > > 3);
                *sptr1 + +  + = ((*sptr2 + +) > > 3);
            }
            break;
        case 2:
            while (--count) {
                *sptr1 + +  + = ((3 * (*sptr2 + +)) > > 4);
                *sptr1 + +  + = ((3 * (*sptr2 + +)) > > 4);
                *sptr1 + +  + = ((3 * (*sptr2 + +)) > > 4);
                *sptr1 + +  + = ((3 * (*sptr2 + +)) > > 4);
            }
            break;
        case 3:
            while (--count) {
                *sptr1 + +  + = ((*sptr2 + +) > > 2);
                *sptr1 + +  + = ((*sptr2 + +) > > 2);
                *sptr1 + +  + = ((*sptr2 + +) > > 2);
                *sptr1 + +  + = ((*sptr2 + +) > > 2);
            }
            break;
        case 4:
            while (--count) {
                *sptr1 + +  + = ((5 * (*sptr2 + +)) > > 4);
                *sptr1 + +  + = ((5 * (*sptr2 + +)) > > 4);
                *sptr1 + +  + = ((5 * (*sptr2 + +)) > > 4);
                *sptr1 + +  + = ((5 * (*sptr2 + +)) > > 4);
            }
        break;
    case 5:
        while (--count) {
            *sptr1++ += ((3 * (*sptr2++)) >> 3);
            *sptr1++ += ((3 * (*sptr2++)) >> 3);
            *sptr1++ += ((3 * (*sptr2++)) >> 3);
            *sptr1++ += ((3 * (*sptr2++)) >> 3);
        }
        break;
    case 6:
        while (--count) {
            *sptr1++ += ((7 * (*sptr2++)) >> 4);
            *sptr1++ += ((7 * (*sptr2++)) >> 4);
            *sptr1++ += ((7 * (*sptr2++)) >> 4);
            *sptr1++ += ((7 * (*sptr2++)) >> 4);
        }
        break;
    case 7:
        while (--count) {
            *sptr1++ += ((*sptr2++) >> 1);
            *sptr1++ += ((*sptr2++) >> 1);
            *sptr1++ += ((*sptr2++) >> 1);
            *sptr1++ += ((*sptr2++) >> 1);
        }
        break;
    case 8:
        while (--count) {
            long     tmp;
            tmp = *sptr2++;
            *sptr1++ += ((8 * tmp + tmp) >> 4);
            tmp = *sptr2++;
            *sptr1++ += ((8 * tmp + tmp) >> 4);
            tmp = *sptr2++;
            *sptr1++ += ((8 * tmp + tmp) >> 4);
            tmp = *sptr2++;
            *sptr1++ += ((8 * tmp + tmp) >> 4);
        }
        break;
    case 9:
        while (--count) {
            *sptr1++ += ((5 * (*sptr2++)) >> 3);
            *sptr1++ += ((5 * (*sptr2++)) >> 3);
            *sptr1++ += ((5 * (*sptr2++)) >> 3);
            *sptr1++ += ((5 * (*sptr2++)) >> 3);
        }
        break;
    case 10:
        while (--count) {
            long tmp;
            tmp = *sptr2++;
```

```
                *sptr1++ += (((tmp << 3) + 3 * tmp) >> 4);
                tmp = *sptr2++;
                *sptr1++ += (((tmp << 3) + 3 * tmp) >> 4);
                tmp = *sptr2++;
                *sptr1++ += (((tmp << 3) + 3 * tmp) >> 4);
                tmp = *sptr2++;
                *sptr1++ += (((tmp << 3) + 3 * tmp) >> 4);
            }
            break;
        case 11:
            while (--count) {
                *sptr1++ += ((3 * (*sptr2++)) >> 2);
                *sptr1++ += ((3 * (*sptr2++)) >> 2);
                *sptr1++ += ((3 * (*sptr2++)) >> 2);
                *sptr1++ += ((3 * (*sptr2++)) >> 2);
            }
            break;
        case 12:
            while (--count) {
                long tmp;
                tmp = *sptr2++;
                *sptr1++ += (((tmp << 4) - 3 * tmp) >> 4);
                tmp = *sptr2++;
                *sptr1++ += (((tmp << 4) - 3 * tmp) >> 4);
                tmp = *sptr2++;
                *sptr1++ += (((tmp << 4) - 3 * tmp) >> 4);
                tmp = *sptr2++;
                *sptr1++ += (((tmp << 4) - 3 * tmp) >> 4);
            }
            break;
        case 13:
            while (--count) {
                *sptr1++ += ((7 * (*sptr2++)) >> 3);
                *sptr1++ += ((7 * (*sptr2++)) >> 3);
                *sptr1++ += ((7 * (*sptr2++)) >> 3);
                *sptr1++ += ((7 * (*sptr2++)) >> 3);
            }
            break;
        case 14:
            while (--count) {
                long tmp;
                tmp = *sptr2++;
                *sptr1++ += (((tmp << 4) - tmp) >> 4);
                tmp = *sptr2++;
                *sptr1++ += (((tmp << 4) - tmp) >> 4);
                tmp = *sptr2++;
                *sptr1++ += (((tmp << 4) - tmp) >> 4);
                tmp = *sptr2++;
                *sptr1++ += (((tmp << 4) - tmp) >> 4);
            }
```

```
                break;
            case 15:
                while (--count) {
                    *sptr1++ += *sptr2++;
                    *sptr1++ += *sptr2++;
                    *sptr1++ += *sptr2++;
                    *sptr1++ += *sptr2++;
                }
                break;
        }
    }

}

short SnDecompress(DecodeBook, ratio, frame_size, min_pitch, bstream, output)
short **DecodeBook, ratio;
unsigned char  *bstream;
short  *output, frame_size, min_pitch;
{
    short    count, SampCount;
    register short    dstate;
    short    vcount;
    short    vsize, cbook_size, bs_size;

    get_compr_pars(ratio, frame_size, &vsize, &cbook_size, &bs_size);

    dstate = *bstream++;
    dstate = (dstate - 128) << 6;

    SampCount = 0;
    while((*bstream & LAST_FRAME_FLAG) == 0)
    {
        expand(DecodeBook, frame_size, vsize, min_pitch,
            (struct frame *)bstream, output, SampCount);
        bstream += bs_size;
        SampCount += frame_size;
    }
    vcount = *bstream - LAST_FRAME_FLAG;
    *bstream = *(bstream + 2 + vcount);
    expand(DecodeBook, frame_size, vsize, min_pitch,
        (struct frame *)bstream, output, SampCount);
    *bstream = vcount + LAST_FRAME_FLAG;
    SampCount += vcount * vsize;

    count = (SampCount >> 1) + 1;
    while (--count) {
        *output++ = dstate = ((IPCONS * dstate) >> 3) + *output;
        *output++ = dstate = ((IPCONS * dstate) >> 3) + *output;
    }
    output -= SampCount;
```

```
  return SampCount;
}

#define    FILTER  state + PFILT_ORDER+1
#define DC_VAL    state + PFILT_ORDER + PFILT_ORDER/2 + 2
void SnSampExpandFilt(short *src, short off, short len,
  char *dest,short *state)
{
  short         input, temp;
  long          acc;
  register short dc = *(DC_VAL);
  register short *sptr1, *sptr2;

  src + = off;
  len+ +;
  sptr1 = state;
  sptr2 = state + PFILT_ORDER;
  while (--len) {
     input = *src+ + - dc;
     dc + = input > > 5;

     temp = input + *sptr1+ +; /* (state[0] + state[8]) * filter[0] */
     acc = temp * *(FILTER);

     temp = *--sptr2 + *sptr1+ +;  /* (state[1] + state[7]) * filter[1] */
     acc + = temp * *(FILTER+1);

     temp = *--sptr2 + *sptr1+ +;  /* (state[2] + state[6]) * filter[2] */
     acc + = temp * *(FILTER+2);

     temp = *--sptr2 + *sptr1+ +;  /* (state[3] + state[5]) * filter[3] */
     acc + = temp * *(FILTER+3);

     acc + = *sptr1 * *(FILTER+4); /* state[4] * filter[4] */

     if (acc > 0)
     {
        temp = (acc + (257 < < 20)) > > 21;
        if (temp > 255)
           temp = 255;
     }
     else
     {
        temp = (acc + (255 < < 20)) > > 21;
        if (temp < 0)
           temp = 0;
     }
     *dest+ + = temp;
```

```
        sptr1 -= 4;
        sptr2 -= 4;
        *sptr1++ = *sptr2++;    /* state[0] = state[1] */
        *sptr1++ = *sptr2++;    /* state[1] = state[2] */
        *sptr1++ = *sptr2++;    /* state[2] = state[3] */
        *sptr1++ = *sptr2++;    /* state[3] = state[4] */
        *sptr1++ = *sptr2++;    /* state[4] = state[5] */
        *sptr1++ = *sptr2++;    /* state[5] = state[6] */
        *sptr1++ = *sptr2++;    /* state[6] = state[7] */
        *sptr1 = input;         /* state[7] = input */
        sptr1 -= 7;
    }
    *(DC_VAL) = dc;
}
```

III. BLENDING MODULE

```
/* A module for blending two diphones */

typedef struct {
   short lptr, pitch;
   short weight, weight_inc;
} bstate;

void SnBlend(pitchp lp, pitchp rp, short cur_tot, short tot,
   short type, bstate *bs)
{
#pragma unused (tot)

   short    count;
   short    *ptr1, *ptr2;

   if (type = = VOICED)
   {
        if (cur_tot)
             return;
        {
             short    weight;
             long     min_amdf;
             short    best_lag = 0, lag;
             short    window_size;
             short    weight_inc;

             /* First replicate the left pitch period */
             ptr1 = lp->bufp;
             ptr2 = ptr1 + lp->olen;
             count = lp->olen + 1;
             while (--count)
                  *ptr2+ + = *ptr1+ +;

             /* Smooth the discontinuity */
             {
                  register short en, e2;

                  en = lp->bufp[2] +
                       3 * (lp->bufp[0] - lp->bufp[1]) - lp->bufp[lp->olen - 1];

                  e2 = lp->bufp[0] - lp->bufp[lp->olen - 1];


                  if (en * en > e2 * e2)
                       en = e2;
```

```
    ptr2 = lp->bufp + lp->olen;
    count = (lp->olen > > 1) + 1;
    while (--count)
    {
        *--ptr2 + = en;
        en = (((en < < 4) - en) > > 4);
    }
}

min_amdf = LARGE_NUM;

window_size = rp->olen;
if (lp->olen < rp->olen)
    window_size = lp->olen;

lag = rp->olen;
while (--lag)
{
    long amdf = 0;
    ptr1 = rp->bufp;
    ptr2 = lp->bufp + lag;
    count = ((window_size+3) > > 2) + 1;
    while (--count)
    {
        short tmp;
        tmp = (*ptr1 - *ptr2);
        if (tmp > 0)
            amdf + = tmp;
        else
            amdf -= tmp;
        ptr1 + = 4;
        ptr2 + = 4;
    }
    if (amdf < min_amdf)
    {
        best_lag = lag;
        min_amdf = amdf;
    }
}

bs->pitch = lp->olen;
/* Update left buffer */
if (best_lag < (lp->olen > > 1))
{
    /* Add best_lag samples to the length of left pulse*/
    lp->olen + = best_lag;
}
else
{
    /* Delete a few samples from the left pulse */
```

```
                lp->olen = best_lag;
            }
            bs->lptr = best_lag;
            weight_inc = 32767/ window_size;
            weight = 32767 - weight_inc;

            ptr1 = rp->bufp;
            ptr2 = lp->bufp + bs->lptr;
            count = window_size+1;
            while (--count)
            {
                *ptr1++ += (((short) (*ptr2++ - *ptr1) * weight) >> 15);
                weight -= weight_inc;
            }
        }
    }
    else
    {
        register short    delta;

        /* Just blend 15 samples */
        ptr2 = lp->bufp + lp->olen - 15;
        ptr1 = rp->bufp;
/*
        for (i = 1; i < 16; i++)
        {
            *ptr1 = *ptr2 + (i * (*ptr1 - *ptr2)) >> 4;
            ptr1++;
            ptr2++;
        }
*/
        delta = *ptr1 - *ptr2;
        *ptr1++ = *ptr2++ + (delta >> 4);

        delta = *ptr1 - *ptr2;
        *ptr1++ = *ptr2++ + ((delta) >> 3);

        delta = *ptr1 - *ptr2;
        *ptr1++ = *ptr2++ + ((3 * delta) >> 4);

        delta = *ptr1 - *ptr2;
        *ptr1++ = *ptr2++ + (delta >> 2);

        delta = *ptr1 - *ptr2;
        *ptr1++ = *ptr2++ + ((5 * delta) >> 4);

        delta = *ptr1 - *ptr2;
        *ptr1++ = *ptr2++ + ((3 * delta) >> 8);

        delta = *ptr1 - *ptr2;
        *ptr1++ = *ptr2++ + ((7 * delta) >> 4);

        delta = *ptr1 - *ptr2;
        *ptr1++ = *ptr2++ + (delta >> 1);

        delta = *ptr1 - *ptr2;
        *ptr1++ = *ptr2++ + (((delta << 3) + delta) >> 4);

        delta = *ptr1 - *ptr2;
        *ptr1++ = *ptr2++ + ((5 * delta) >> 3);

        delta = *ptr1 - *ptr2;
        *ptr1++ = *ptr2++ + (((delta << 3) + 3 * delta) >> 4);

        delta = *ptr1 - *ptr2;
        *ptr1++ = *ptr2++ + ((3 * delta) >> 2);

        delta = *ptr1 - *ptr2;
        *ptr1++ = *ptr2++ + (((delta << 4) - 3 * delta) >> 4);

        delta = *ptr1 - *ptr2;
        *ptr1++ = *ptr2++ + ((7 * delta) >> 3);

        delta = *ptr1 - *ptr2;
        *ptr1 = *ptr2 + (((delta << 4) - delta) >> 4);


        lp->olen -= 15;
    }
}
```

IV. INTONATION ADJUSTMENT MODULE

```
/* A module for deleting a pitch period */
/*
   Pointer src1 points to Left Pitch period
   Pointer src2 points to Right Pitch period
   Pointer dst points to Resulting Pitch period
   len = length of the pitch periods
*/
skip_pulses(short *src1, short *src2, short *dst, short len)
{
   short i;
   register short    weight, cweight;

   i = len+1;
   weight = cweight = 32767/i;
   while (--i)
   {
       *dst+ + = *src1+ + + (((short) (*src2+ + - *src1) * cweight) > > 15);
       cweight + = weight;
   }
}

/* A module for Inserting a pitch period */
/*
   Locn buffer[curbeg] points to Left Pitch period
   Locn buffer[curbeg+curlen] points to Right Pitch period
   Pointer dst points to Resulting Pitch period
   curlen = length of the pitch periods
*/
insert_pulse(short *buffer, short *dst, short curlen, short curbeg)
{
   short    weight, cweight, count;
   short    *src1, *src2;

   src1 = buffer + curbeg;
   src2 = buffer + curbeg + curlen;
   weight = 32767 / curlen;
   cweight = weight;
   count = curlen + 1;
   while (--count)
   {
       *dst+ + = *src1+ + = *src2+ + + (((short) (*src1 - *src2) * cweight) > >
15);
       cweight + = weight;
   }
}

   /* This module is used to change pitch information in the concatenated speech */
   // This routine depends on the desired length (deslen) being at least half
   // and no more than twice the actual length (len).

   void SnChangePitch(short *buf, short *next, short len, short deslen,short lvoc,short
rvoc,short dosmooth)
   {
   #pragma unused(rvoc, dosmooth)
       short    delta;
       short    count;
       short    *bptr, *aptr;
       short    weight, weight_inc;
       if (!lvoc | | (deslen = = len)) return;

       if (deslen > len)
       {
           /* Increase Pitch period */
           delta = deslen - len;
           bptr = buf + len;
           aptr = buf + deslen;
           count = delta + 1;
           while (--count)
               *--aptr = *--bptr;

           count = len - delta + 1;
           weight = weight_inc = 32767 / count;
           while (--count)
           {
               register short tmp2;
               tmp2 = (*--aptr - *--bptr);
               *aptr = *bptr + ((tmp2 * weight) > > 15);
               weight + = weight_inc;
           }
           return;
       }
       {
           /* Shorten Pitch Period */
           short wsize;

           delta = len - deslen;
           wsize = 2 * delta;

           if (wsize > deslen)
               wsize = deslen;

           weight_inc = 32767 / (wsize + 1);
           weight = weight_inc;
           aptr = buf + deslen;
           bptr = buf + len  - wsize;
           count = wsize - delta + 1;
```

```
            while (--count)
            {
                *bptr++ += (((short) (*aptr++ - *bptr) * weight ) >> 15);
                weight += weight_inc;
            }
            aptr = buf + deslen;
            bptr = next;
            count = delta + 1;
            weight = 32767 - weight;
            while (--count)
            {
                *bptr++ += (((short) (*aptr++ - *bptr) * weight ) >> 15);
                weight -= weight_inc;
            }
        }
    }
```

Claims

1. An apparatus for adjusting an intonation of a sound wherein the sound is specified by a sequence of frames each comprising a set of digital samples, the apparatus comprising:

    means for receiving a set of intonation control signals that indicate a pitch adjustment and a duration adjustment to the sound;
    a buffer that stores the sequence of frames;
    intonation control means that generates an intonation adjusted sequence of frames by accessing a block of one or more frames of the sequence of frames from the buffer and by generating a modified block in response to the intonation control signals and by inserting the modified block into the sequence of frames;

    characterized by comprising means for applying a first weighting function to the block emphasizing the beginning segment to generate a first vector and means for applying a second weighting function to the block emphasizing the ending segment to generate a second vector, and means for combining the first vector with the second vector to generate the modified block, such that the intonation control means minimizes a discontinuity between a beginning segment and an ending segment of the block and a pair of adjacent frames in the intonation adjusted sequence of frames.

2. The apparatus of claim 1, wherein the intonation control signals indicate a change in a nominal length of a specified frame of the sequence of frames to indicate the pitch adjustment and indicate a change in a number of frames in the sequence of frames to indicate the duration adjustment.

3. The apparatus of claim 2, wherein the intonation control means includes pitch lowering means for increasing a length N of the specified frame by an amount equal to Δ samples wherein the block of one or more frames consists of the specified frame, the pitch lowering means including said means for applying a first weighting function and said means for applying a second weighting function, and wherein said means for combining comprises means for combining the first vector with the second vector shifted by Δ samples to generate the modified block having a length N+Δ.

4. The apparatus of claim 2, wherein the intonation control means includes pitch raising means for decreasing a length N of the specified frame by an amount equal to Δ samples wherein the block of one or more frames consists of the specified frame and a next frame having a length NR in the sequence of frames, the pitch raising means including said means for applying a first weighting function and said means for applying a second weighting function, and wherein said means for combining comprises means for combining the first vector with the second vector shifted by Δ samples to generate a shortened frame with the next frame to generate the modified block having a

length N-Δ+NR.

5. The apparatus of claim 2, wherein the intonation control means includes duration shortening means for modifying the block to reduce the number of frames in the sequence of frames wherein the block consist of a pair of sequential frames having lengths NL and NR respectively, the duration shortening means including said means for applying a first weighting function and said means for applying a second weighting function, and wherein said means for combining comprises means for combining the first vector with the second vector to generate the modified block having the length NL or the length NR.

6. The apparatus of claim 2, wherein the intonation control means includes duration lengthening means for modifying the block to increase the number of frames in the sequence of frames wherein the block consists of a pair of left and right sequential frames having lengths NL and NR respectively, the duration lengthening means including said means for applying a first weighting function and said means for applying a second weighting function, and wherein said means for combining comprises means for combining the first vector with the second vector to generate a new frame and means for concatenating the left frame, the new frame, and the right frame to generate the modified block.

Patentansprüche

1. Vorrichtung zur Einstellung einer Intonation eines Tons bzw. Klangs, bei welcher der Ton durch eine Folge von Rahmen spezifiziert ist, welche jeweils einen Satz digitaler Proben umfassen, wobei die Vorrichtung aufweist:

   - Mittel zum Empfang eines Satzes von Intonationssteuersignalen, welche eine Tonhöhen-Einstellung und eine Dauer-Einstellung des Tones anzeigen;
   - einen Puffer, welcher die Folge von Rahmen speichert;
   - Intonationssteuermittel, die eine intonationseingestellte Folge von Rahmen generieren durch Zugriff auf einen Block von einem oder einer Anzahl von Rahmen der Folge von Rahmen aus dem Puffer und durch Generierung eines modifizierten Blocks ansprechend auf die Intonationssteuersignale und durch Einfügen des modifizierten Blocks in die Folge der Rahmen;

   dadurch gekennzeichnet,
   daß sie Mittel zur Anwendung einer ersten Gewichtungsfunktion auf den Block, welche das Anfangssegment betont, zur Generierung eines ersten Vektors und Mittel zur Anwendung einer zweiten Gewichtungsfunktion auf den Block, welche das Endsegment betont, zur Generierung eines zweiten Vektors, und Mittel zur Kombination des ersten Vektors mit dem zweiten Vektor zur Generierung des modifizierten Blocks derart, daß die Intonationssteuermittel eine Diskontinuität zwischen einem Anfangssegment und einem Endsegment des Blocks und einem Paar von benachbarten Rahmen in der intonationseingestellten Folge von Rahmen minimieren, aufweist.

2. Vorrichtung nach Anspruch 1, bei welcher die Intonationssteuersignale eine Änderung in der nominalen Länge eines spezifizierten Rahmens der Folge von Rahmen zur Anzeige der Einstellung der Tonhöhe anzeigen, und eine Änderung in einer Anzahl der Rahmen in der Folge von Rahmen zur Anzeige der Einstellung der Dauer anzeigen.

3. Vorrichtung nach Anspruch 2, bei welcher die Intonationssteuermittel Tonhöhensenkungsmittel zur Erhöhung einer Länge N des spezifizierten Rahmens um einen Betrag, welcher gleich Δ Proben ist, beinhalten, wobei der Block von einem oder einer Anzahl von Rahmen aus dem spezifizierten Rahmen besteht, wobei die Tonhöhensenkungsmittel die Mittel zum Anwenden einer ersten Gewichtungsfunktion und die Mittel zur Anwendung einer zweiten Gewichtungsfunktion beinhalten, und wobei die Mittel zur Kombination Mittel zur Kombination des ersten Vektors mit dem um Δ Proben versetzten zweiten Vektor zur Generierung des modifizierten Blocks mit einer Länge N+Δ aufweisen.

4. Vorrichtung nach Anspruch 2, bei welcher die Intonationssteuermittel Tonhöhenerhöhungsmittel zur Verringerung einer Länge N des spezifizierten Rahmens um einen Betrag, welcher gleich Δ Proben ist, beinhalten, wobei der Block von einem oder einer Anzahl von Rahmen aus dem spezifizierten Rahmen und einem nächsten Rahmen mit einer Länge NR in der Folge von Rahmen besteht, wobei die Tonhöhenerhöhungsmittel die Mittel zur Anwendung einer ersten Gewichtungsfunktion und die Mittel zur Anwendung einer zweiten Gewichtungsfunktion beinhalten, und wobei die Mittel zur Kombination Mittel zur Kombination des ersten Vektors mit dem um Δ Proben versetzten zweiten Vektor zur Generierung eines verkürzten Rahmens mit dem nächsten Rahmen zur Generierung

des modifizierten Blocks mit einer Länge N-Δ+NR aufweisen.

5. Vorrichtung nach Anspruch 2, bei welcher die Intonationssteuermittel Dauer-Verkürzungsmittel zur Modifizierung des Blocks zur Reduzierung der Anzahl von Rahmen in der Folge von Rahmen beinhalten, wobei der Block aus einem Paar von sequentiellen Rahmen mit Längen NL bzw. NR besteht, wobei die Dauer-Verkürzungsmittel die Mittel zur Anwendung einer ersten Gewichtungsfunktion und die Mittel zur Anwendung einer zweiten Gewichtungsfunktion beinhalten, und wobei die Mittel zur Kombination Mittel zur Kombination des ersten Vektors mit dem zweiten Vektor zur Generierung des modifizierten Blocks mit der Länge NL oder der Länge NR aufweisen.

6. Vorrichtung nach Anspruch 2, bei welcher die Intonationssteuermittel Dauer-Verlängerungsmittel zur Modifizierung des Blocks zur Erhöhung der Anzahl der Rahmen in der Folge von Rahmen beinhalten, wobei der Block aus einem Paar von linken und rechten sequentiellen Rahmen mit Längen NL bzw. NR besteht, wobei die Dauer-Verlängerungsmittel die Mittel zur Anwendung einer ersten Gewichtungsfunktion und die Mittel zur Anwendung einer zweiten Gewichtungsfunktion beinhalten, und wobei die Mittel zur Kombination Mittel zur Kombination des ersten Vektors mit dem zweiten Vektor zur Generierung eines neuen Rahmens und Mittel zur Verkettung des linken Rahmens, eines neuen Rahmens und des rechten Rahmens zur Generierung des modifizierten Blocks aufweisen.

Revendications

1. Dispositif pour régler une intonation d'un son, dans lequel le son est spécifié par une séquence de trames dont chacune comprend un ensemble d'échantillons numériques, le dispositif comprenant :

   des moyens pour recevoir un ensemble de signaux de commande d'intonation, qui indiquent un réglage de la hauteur de son et un réglage de la durée du son;
   un tampon qui mémorise la séquence de trames;
   des moyens de commande d'intonation qui génèrent une séquence de trames dont l'intonation est ajustée, par accès à un bloc d'une ou de plusieurs trames de la séquence de trames à partir du tampon et par production d'un bloc modifié en réponse aux signaux de commande d'intonation et par insertion du bloc modifié dans la séquence de trames;

   caractérisé en ce qu'il comporte des moyens pour appliquer une première fonction de pondération au bloc en accentuant le segment de début pour produire un premier vecteur et des moyens pour appliquer une seconde fonction de pondération au bloc en accentuant le segment de fin pour produire un second vecteur, et des moyens pour combiner le premier vecteur avec le second vecteur pour produire le bloc modifié de sorte que les moyens de commande d'intonation réduisent une discontinuité entre un segment de début et un segment de fin du bloc et un couple de trames adjacentes dans la séquence de trames dont l'intonation est réglée.

2. Dispositif selon la revendication 1, dans lequel les signaux de commande d'intonation indiquent une modification de la longueur nominale d'une trame spécifiée de la séquence de trames pour indiquer le réglage de la hauteur de son et une modification du nombre de trames dans la séquence de trames pour indiquer le réglage de durée.

3. Dispositif selon la revendication 2, dans lequel les moyens de commande d'atténuation incluent des moyens de réduction de la hauteur de son pour accroître une longueur N de la trame spécifiée, d'une quantité égale à Δ échantillons, dans lequel le bloc formé d'une ou plusieurs trames est constitué par la trame spécifiée, les moyens de réduction de la hauteur de son incluant lesdits moyens pour appliquer une première fonction de pondération et lesdits moyens pour appliquer une seconde fonction de pondération, et dans lequel lesdits moyens de combinaison comprennent des moyens pour combiner le premier vecteur au second vecteur décalé de Δ échantillons pour produire le bloc modifié possédant une longueur N + Δ.

4. Dispositif selon la revendication 2, dans lequel les moyens de commande d'intonation incluent des moyens d'augmentation de la hauteur de son pour réduire une longueur N de la trame spécifiée d'une quantité égale à Δ échantillons, dans lequel le bloc d'une ou de plusieurs trames est constitué par la trame spécifiée et une trame suivante possédant une longueur NR dans la séquence de trames, les moyens d'augmentation de la hauteur de son comprenant lesdits moyens pour appliquer une première fonction de pondération et lesdits moyens pour appliquer une seconde fonction de pondération, et dans lequel lesdits moyens de combinaison comprennent des moyens pour combiner le premier vecteur avec le second vecteur décalé de Δ échantillons pour produire une trame raccourcie avec la trame suivante de manière à produire le bloc modifié possédant une longueur N - Δ + NR.

5. Dispositif selon la revendication 2, dans lequel les moyens de commande de l'intonation incluent des moyens de réduction de durée pour modifier le bloc afin de réduire le nombre de trames dans la séquence de trames, dans lequel le bloc est constitué par un couple de trames séquentielles ayant respectivement les longueurs NL et NR, les moyens de réduction de durée incluant lesdits moyens pour appliquer une première fonction de pondération et lesdits moyens pour appliquer une seconde fonction de pondération, et dans lequel lesdits moyens de combinaison comprennent les moyens pour combiner le premier vecteur au second vecteur pour produire le bloc modifié possédant la longueur NL ou la longueur NR.

6. Dispositif selon la revendication 2, dans lequel les moyens de commande de l'intonation incluent des moyens d'accroissement de durée pour modifier le bloc afin d'augmenter le nombre de trames dans la séquence de trames, dans lequel le bloc est constitué par un couple de trames séquentielles gauche et droite ayant respectivement les longueurs NL et NR, les moyens d'accroissement de durée incluant lesdits moyens pour appliquer une première fonction de pondération et lesdits moyens pour appliquer une seconde fonction de pondération, et dans lequel lesdits moyens de combinaison comprennent les moyens pour combiner le premier vecteur au second vecteur pour produire une nouvelle trame et des moyens pour concaténer la trame de gauche, la nouvelle trame et la trame de droite pour produire le bloc modifié.

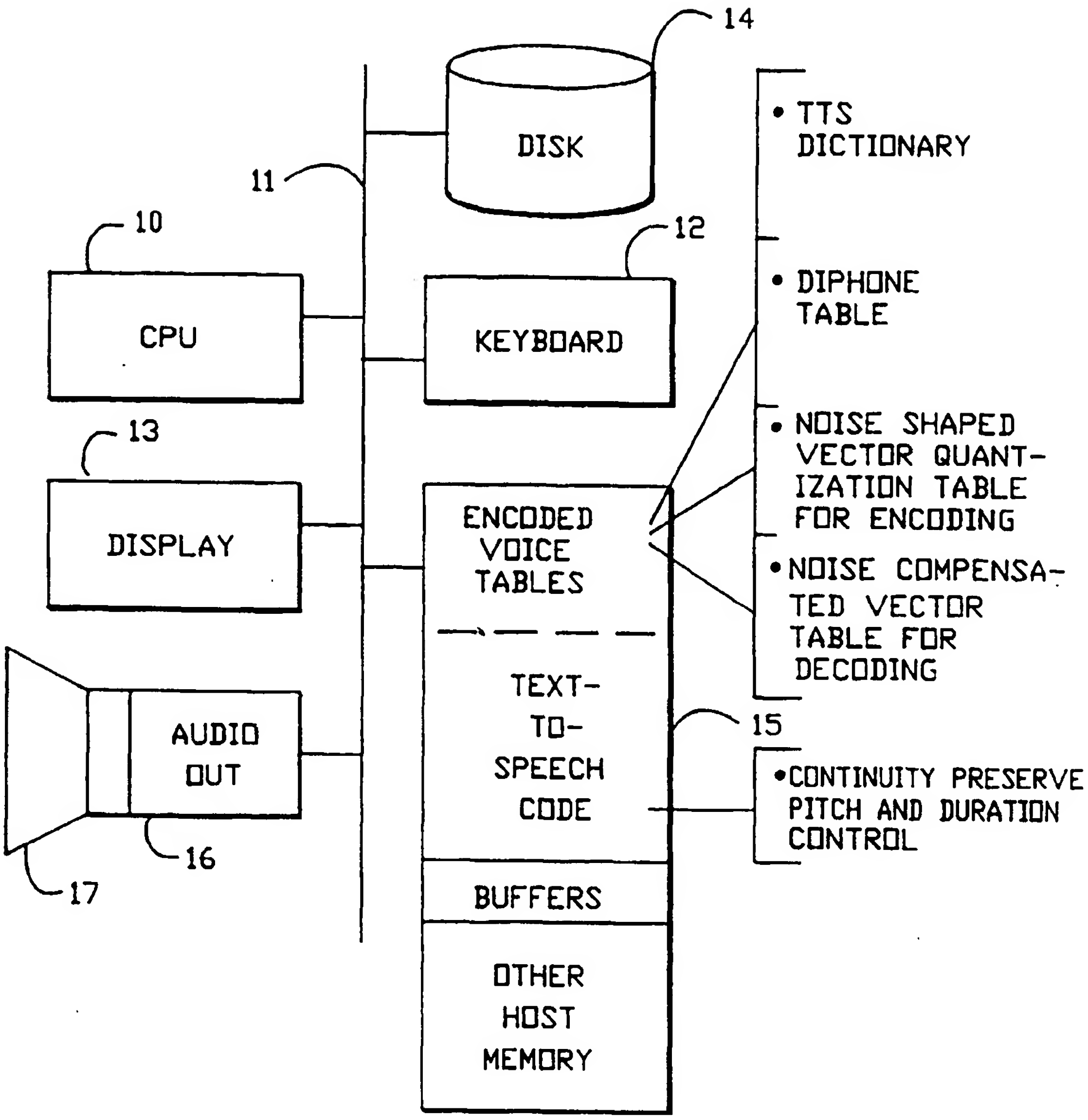
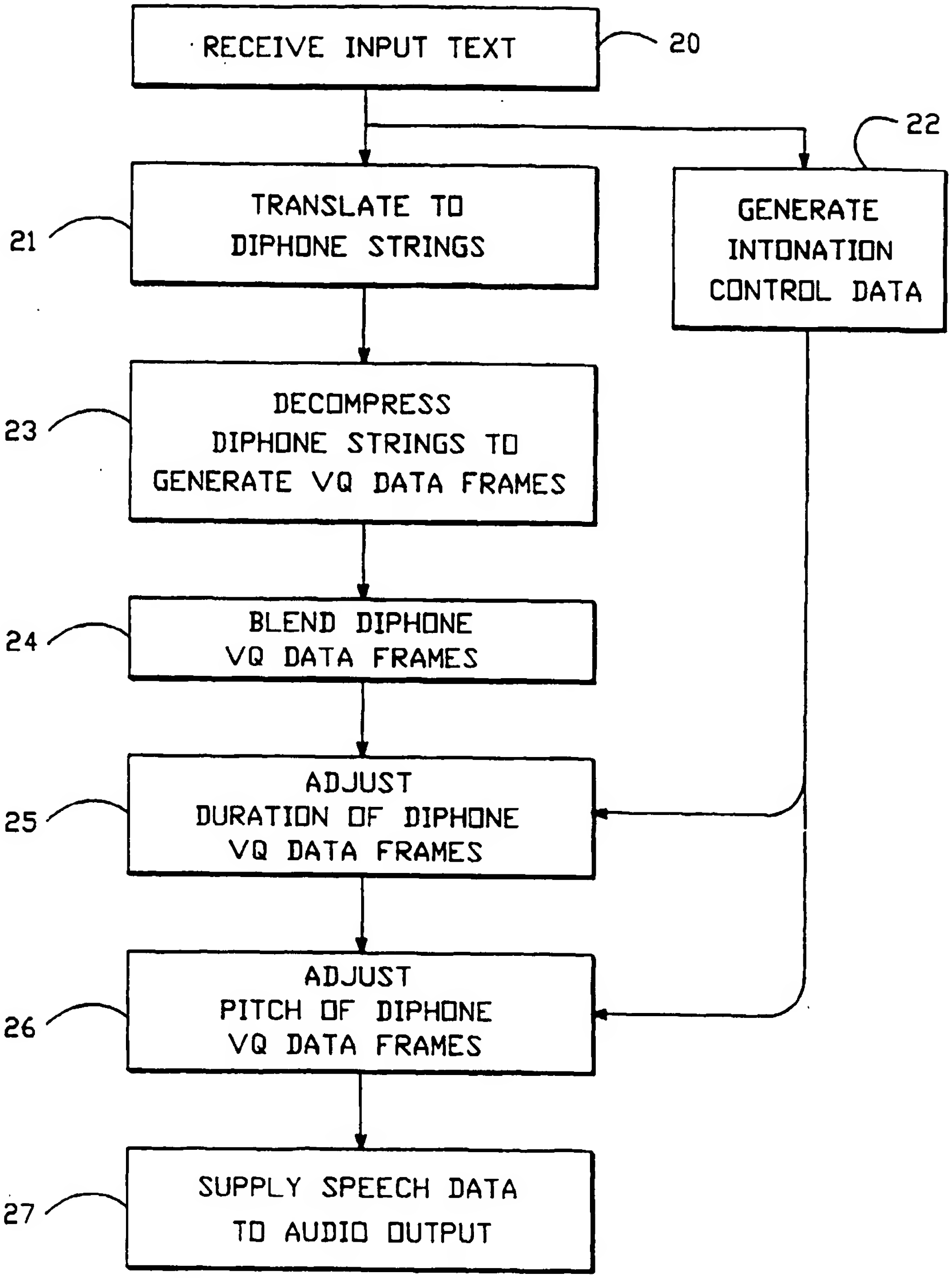

14
DISK
11
10
CPU
12
KEYBOARD
13
DISPLAY
ENCODED VOICE TABLES
TEXT-TO-SPEECH CODE
BUFFERS
OTHER HOST MEMORY
AUDIO OUT
16
17
15
• TTS DICTIONARY
• DIPHONE TABLE
• NOISE SHAPED VECTOR QUANTIZATION TABLE FOR ENCODING
• NOISE COMPENSATED VECTOR TABLE FOR DECODING
•CONTINUITY PRESERVE PITCH AND DURATION CONTROL

FIG.-1


RECEIVE INPUT TEXT
20
TRANSLATE TO DIPHONE STRINGS
21
GENERATE INTONATION CONTROL DATA
22
DECOMPRESS DIPHONE STRINGS TO GENERATE VQ DATA FRAMES
23
BLEND DIPHONE VQ DATA FRAMES
24
ADJUST DURATION OF DIPHONE VQ DATA FRAMES
25
ADJUST PITCH OF DIPHONE VQ DATA FRAMES
26
SUPPLY SPEECH DATA TO AUDIO OUTPUT
27

TEXT - TO - SPEECH CODE

FIG.—2

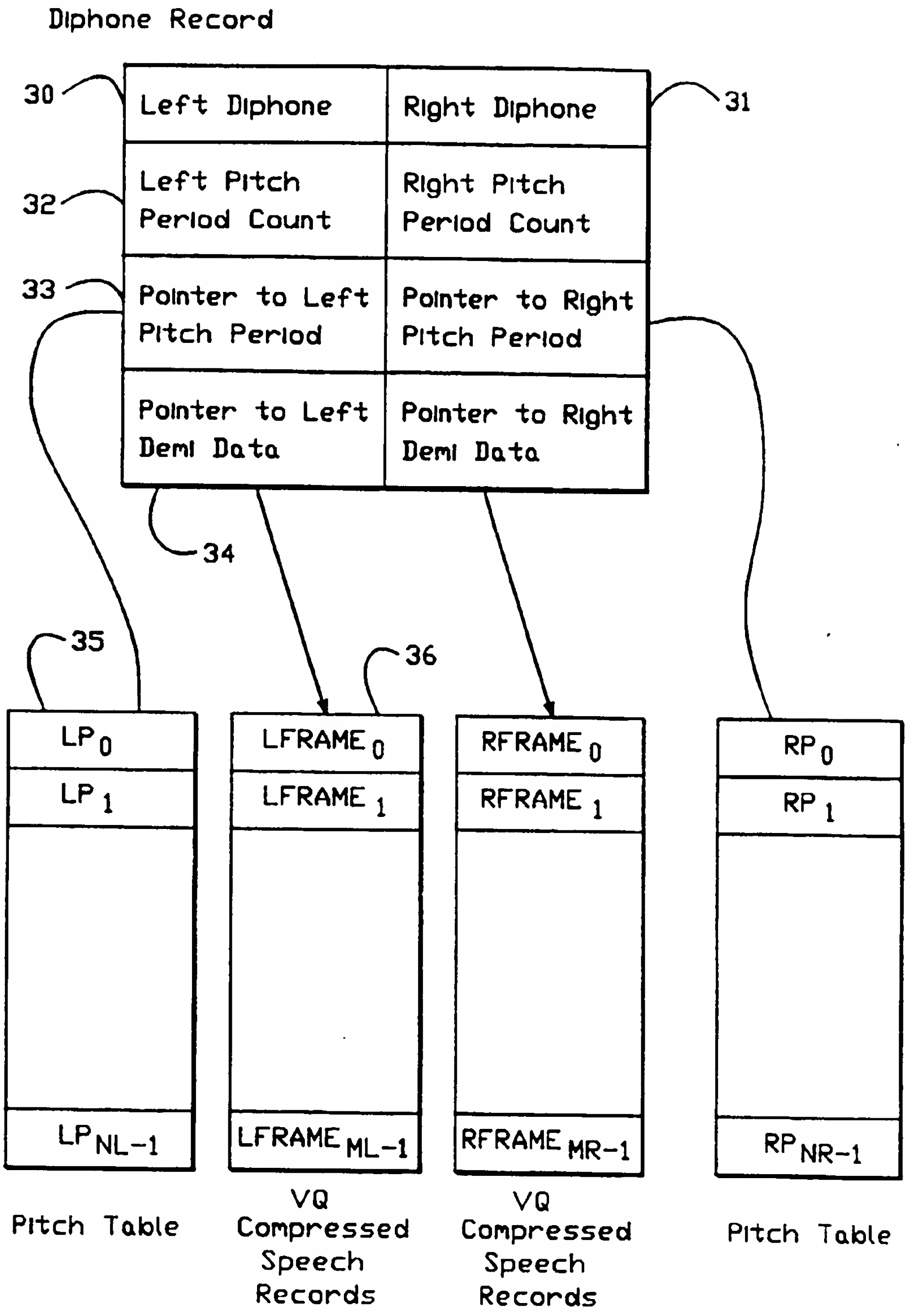

Diphone Record
30
Left Diphone
Right Diphone
31
32
Left Pitch Period Count
Right Pitch Period Count
33
Pointer to Left Pitch Period
Pointer to Right Pitch Period
Pointer to Left Demi Data
Pointer to Right Demi Data
34
35
36
LP0
LP1
LPNL-1
LFRAME0
LFRAME1
LFRAMEML-1
RFRAME0
RFRAME1
RFRAMEMR-1
RP0
RP1
RPNR-1
Pitch Table
VQ Compressed Speech Records
VQ Compressed Speech Records
Pitch Table

FIG.-3

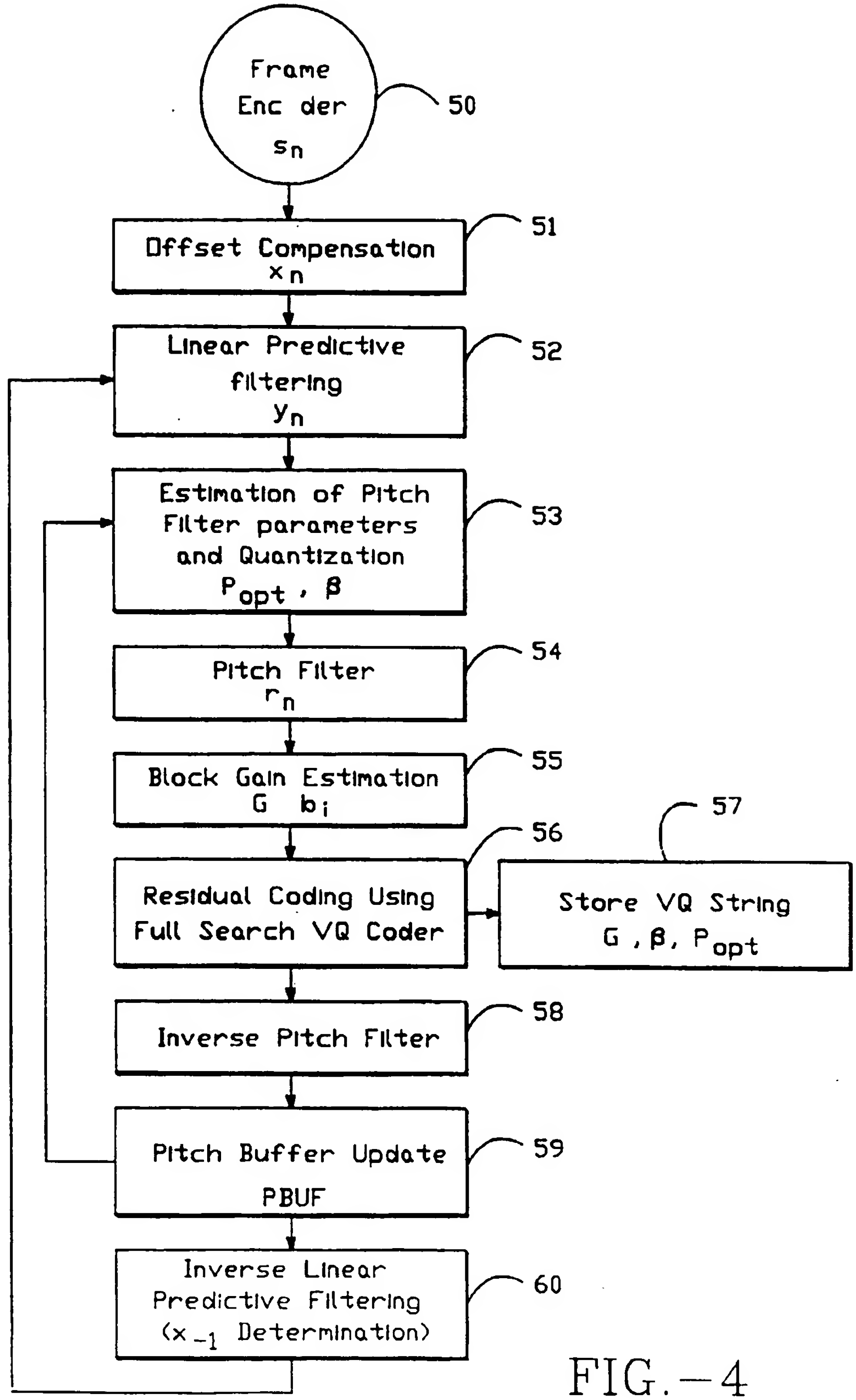

Frame
Enc der
s_n
50
Offset Compensation
x_n
51
Linear Predictive filtering
y_n
52
Estimation of Pitch Filter parameters and Quantization
P_{opt}, β
53
Pitch Filter
r_n
54
Block Gain Estimation
G b_i
55
56
57
Residual Coding Using Full Search VQ Coder
Store VQ String
G, β, P_{opt}
Inverse Pitch Filter
58
Pitch Buffer Update
PBUF
59
Inverse Linear Predictive Filtering
(x_{-1} Determination)
60

FIG.—4

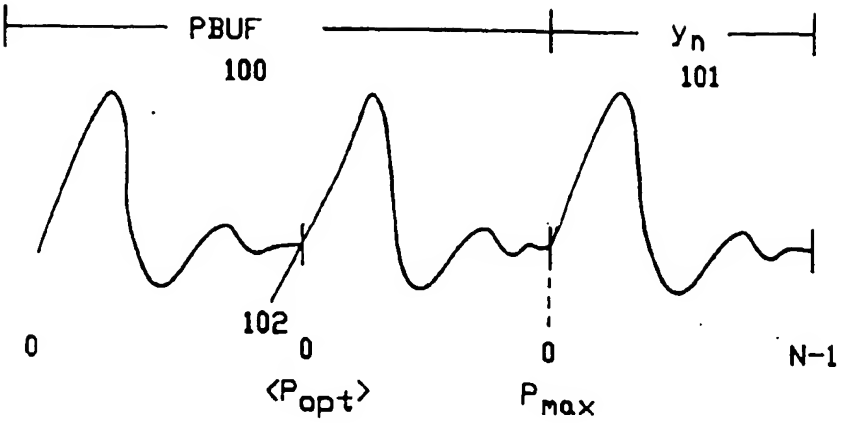

PBUF
100
y_n
101
102
0
0
0
N-1
$\langle P_{opt} \rangle$
P_{max}

FIG.—5

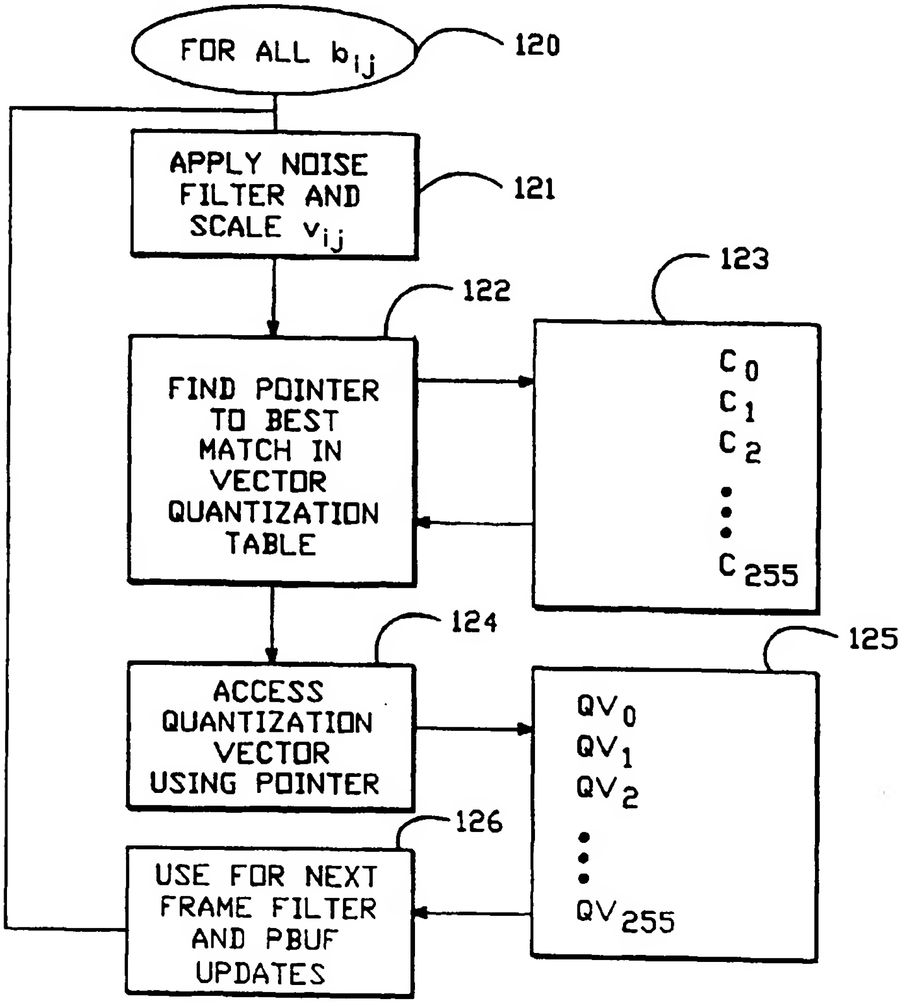

FOR ALL b_{ij}
120
APPLY NOISE FILTER AND SCALE v_{ij}
121
FIND POINTER TO BEST MATCH IN VECTOR QUANTIZATION TABLE
122
123
C_0
C_1
C_2
C_{255}
ACCESS QUANTIZATION VECTOR USING POINTER
124
125
QV_0
QV_1
QV_2
QV_{255}
USE FOR NEXT FRAME FILTER AND PBUF UPDATES
126

FIG.—6

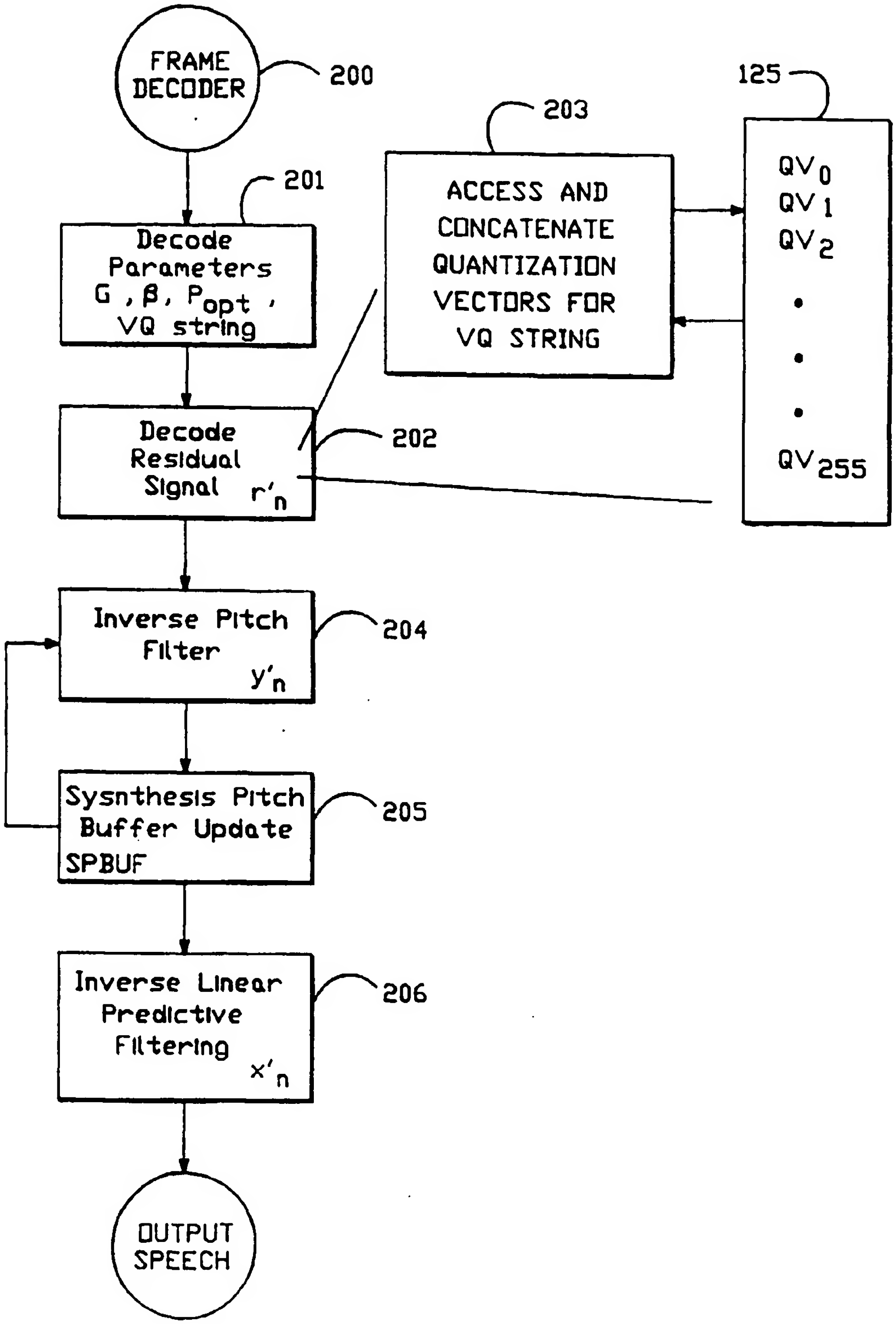

FRAME DECODER
200
Decode Parameters G , β, P_opt , VQ string
201
ACCESS AND CONCATENATE QUANTIZATION VECTORS FOR VQ STRING
203
125
QV0
QV1
QV2
QV255
Decode Residual Signal r'n
202
Inverse Pitch Filter y'n
204
Sysnthesis Pitch Buffer Update SPBUF
205
Inverse Linear Predictive Filtering x'n
206
OUTPUT SPEECH

FIG.-7

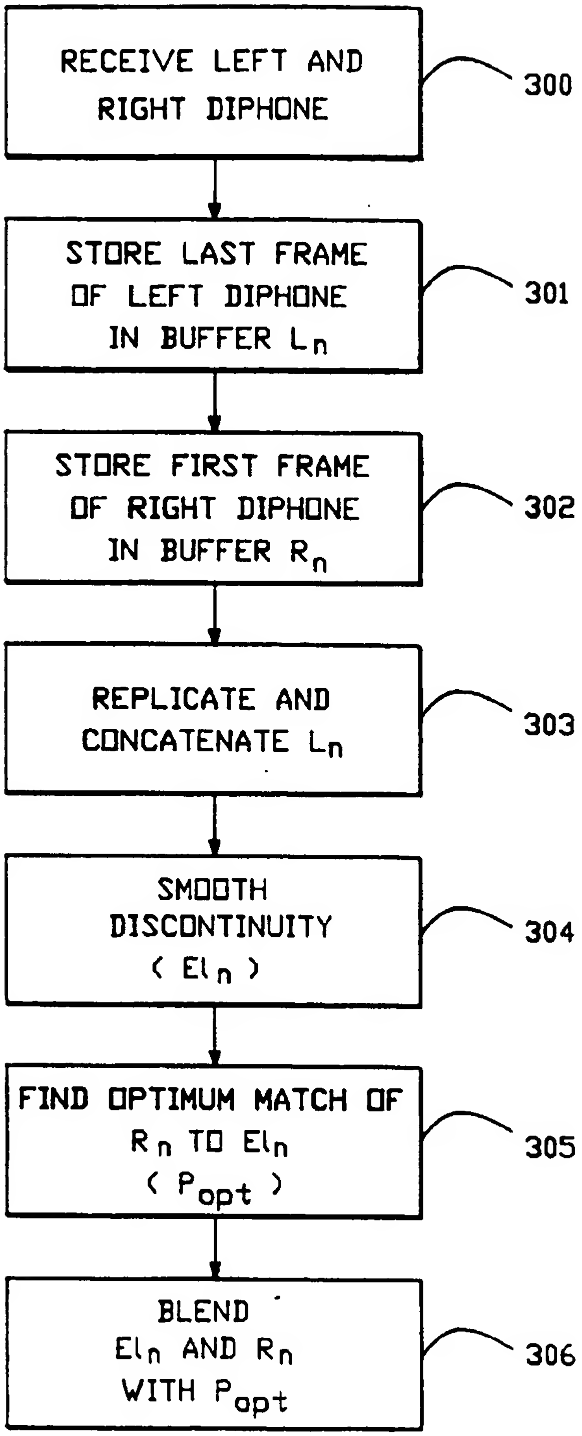

RECEIVE LEFT AND RIGHT DIPHONE
300
STORE LAST FRAME OF LEFT DIPHONE IN BUFFER L_n
301
STORE FIRST FRAME OF RIGHT DIPHONE IN BUFFER R_n
302
REPLICATE AND CONCATENATE L_n
303
SMOOTH DISCONTINUITY (El_n)
304
FIND OPTIMUM MATCH OF R_n TO El_n (P_{opt})
305
BLEND El_n AND R_n WITH P_{opt}
306

FIG.—8

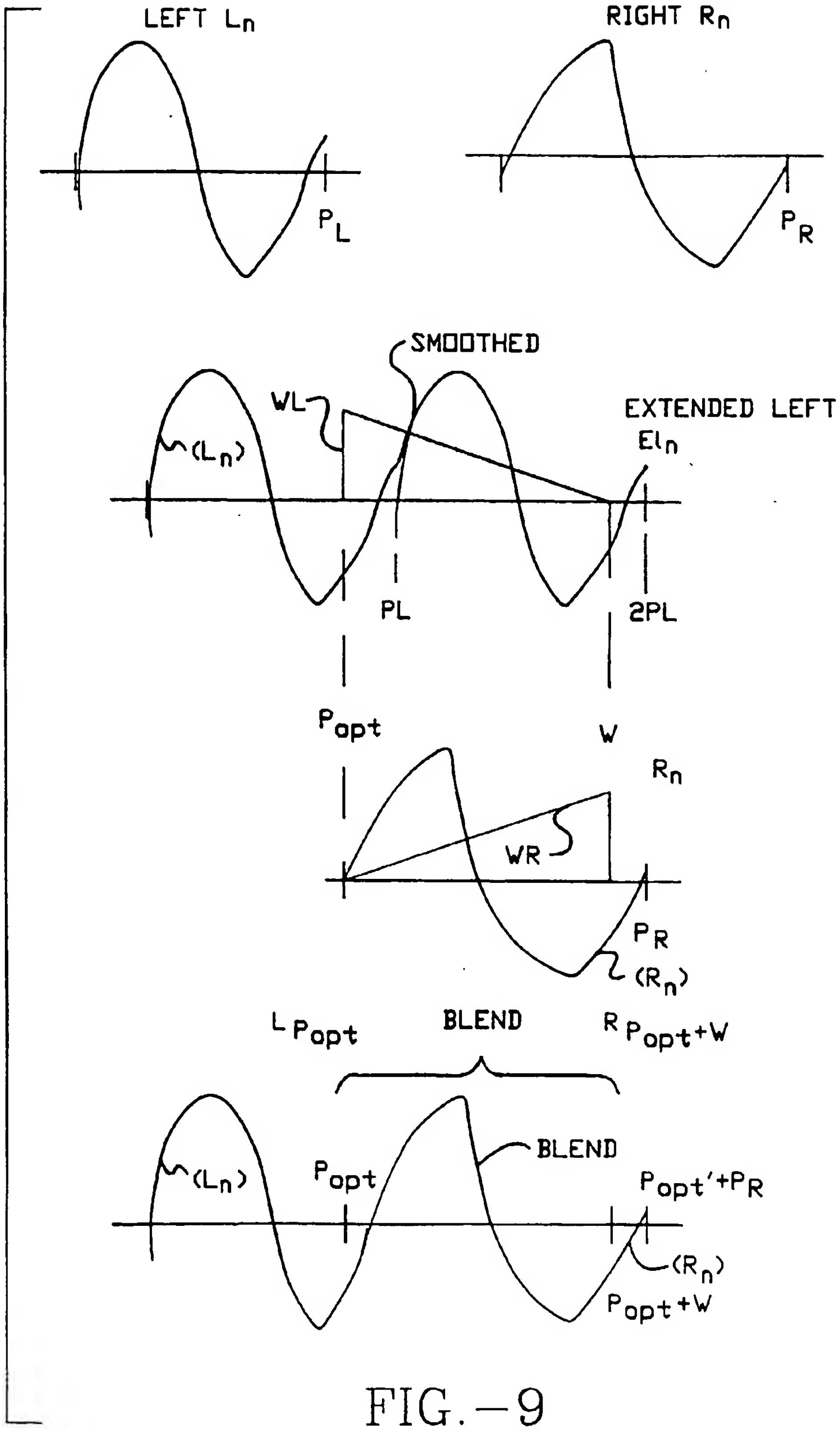
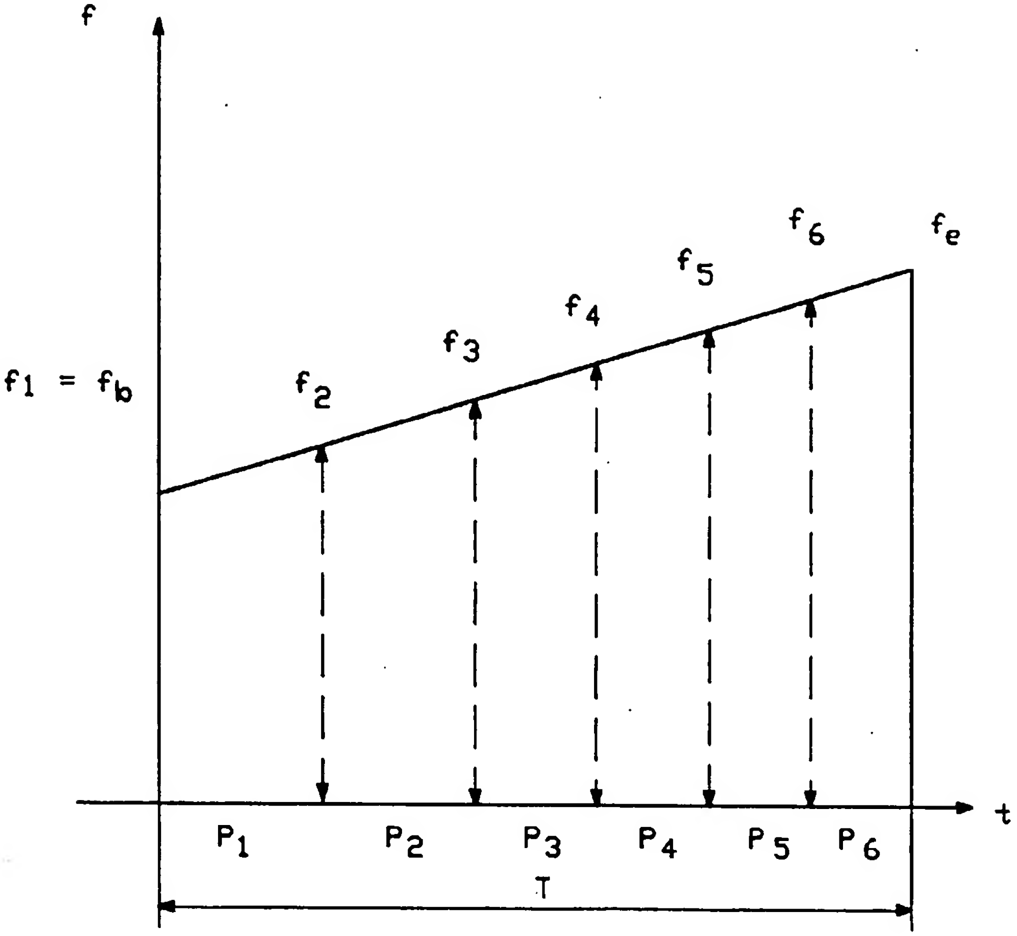

LEFT Ln
RIGHT Rn
PL
PR
SMOOTHED
WL
(Ln)
EXTENDED LEFT
Eln
PL
2PL
Popt
W
Rn
WR
PR
(Rn)
L Popt
BLEND
R Popt+W
(Ln)
Popt
BLEND
Popt'+PR
(Rn)
Popt+W

FIG.—9



NOTES:

T = Desired duration of a phoneme

f_b = Desired Begining Pitch in Hz

f_e = Desired Ending Pitch in Hz

P1, P2, ..., P6 are the desired pitch period in No. of Samples corresponding to the frequencies f1,f2,...f6.

Relationship between Pi and fi:

Pi = Fs/fi, where Fs is the Sampling frequency.

FIG.—10

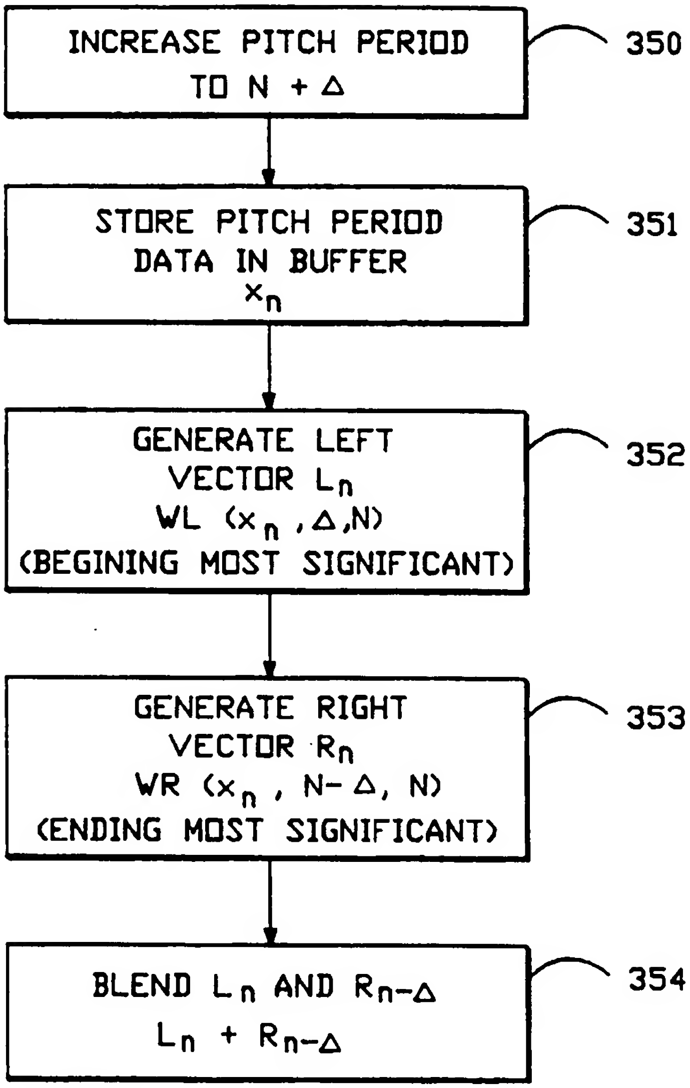

INCREASE PITCH PERIOD
TO N + Δ
350
STORE PITCH PERIOD
DATA IN BUFFER
x_n
351
GENERATE LEFT
VECTOR L_n
WL (x_n ,Δ,N)
(BEGINING MOST SIGNIFICANT)
352
GENERATE RIGHT
VECTOR R_n
WR (x_n , N- Δ, N)
(ENDING MOST SIGNIFICANT)
353
BLEND L_n AND $R_{n-Δ}$
L_n + $R_{n-Δ}$
354

FIG.—11

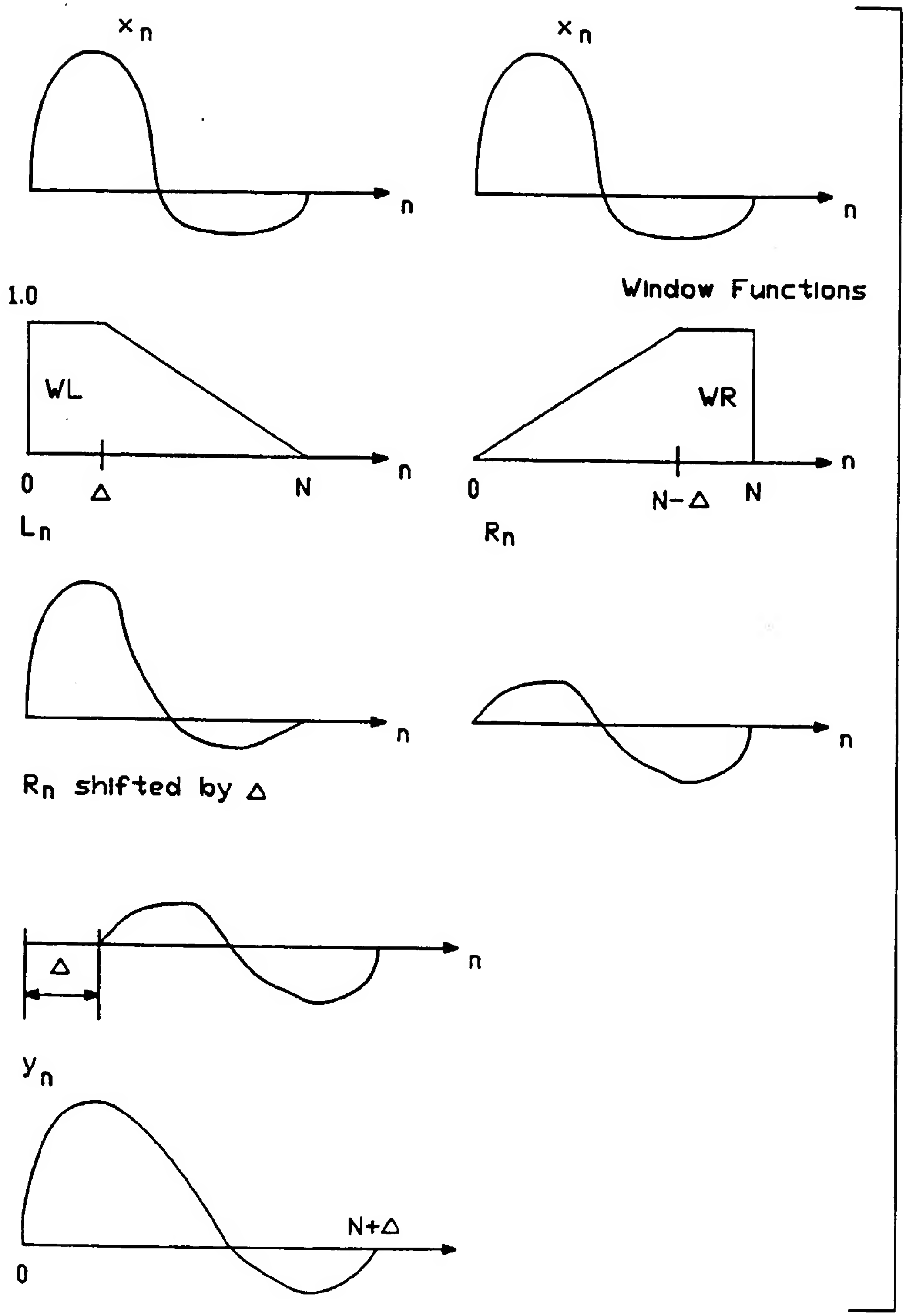

x n
n
x n
n
Window Functions
1.0
WL
n
0
Δ
N
L n
WR
n
0
N−Δ
N
R n
n
R n shifted by Δ
n
Δ
n
y n
N+Δ
0

FIG.-12

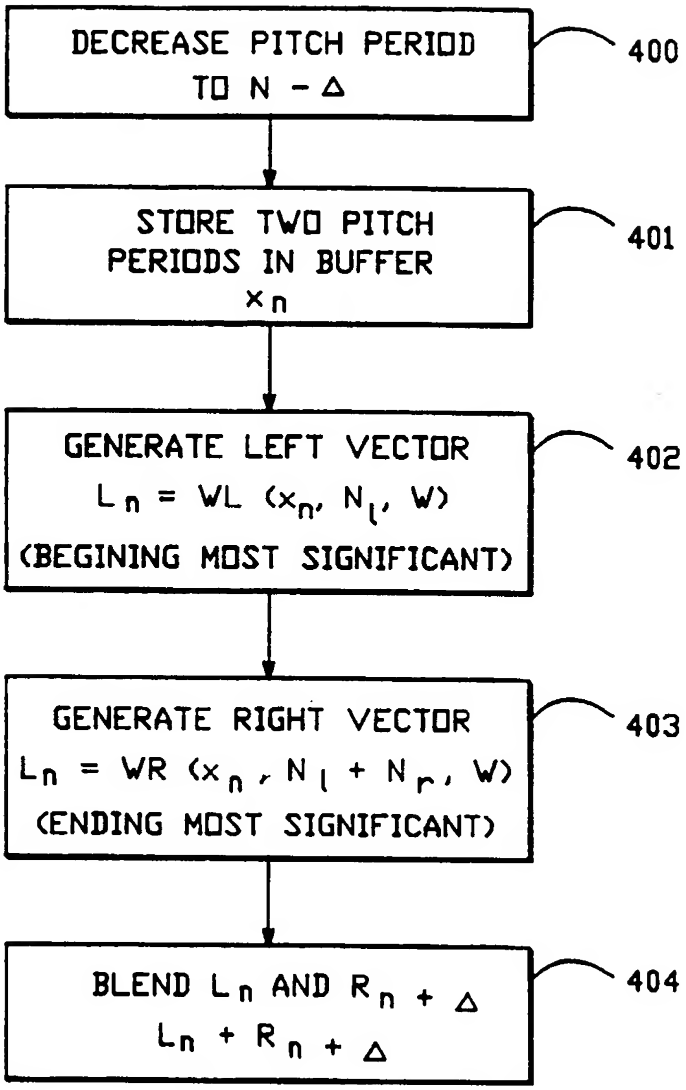
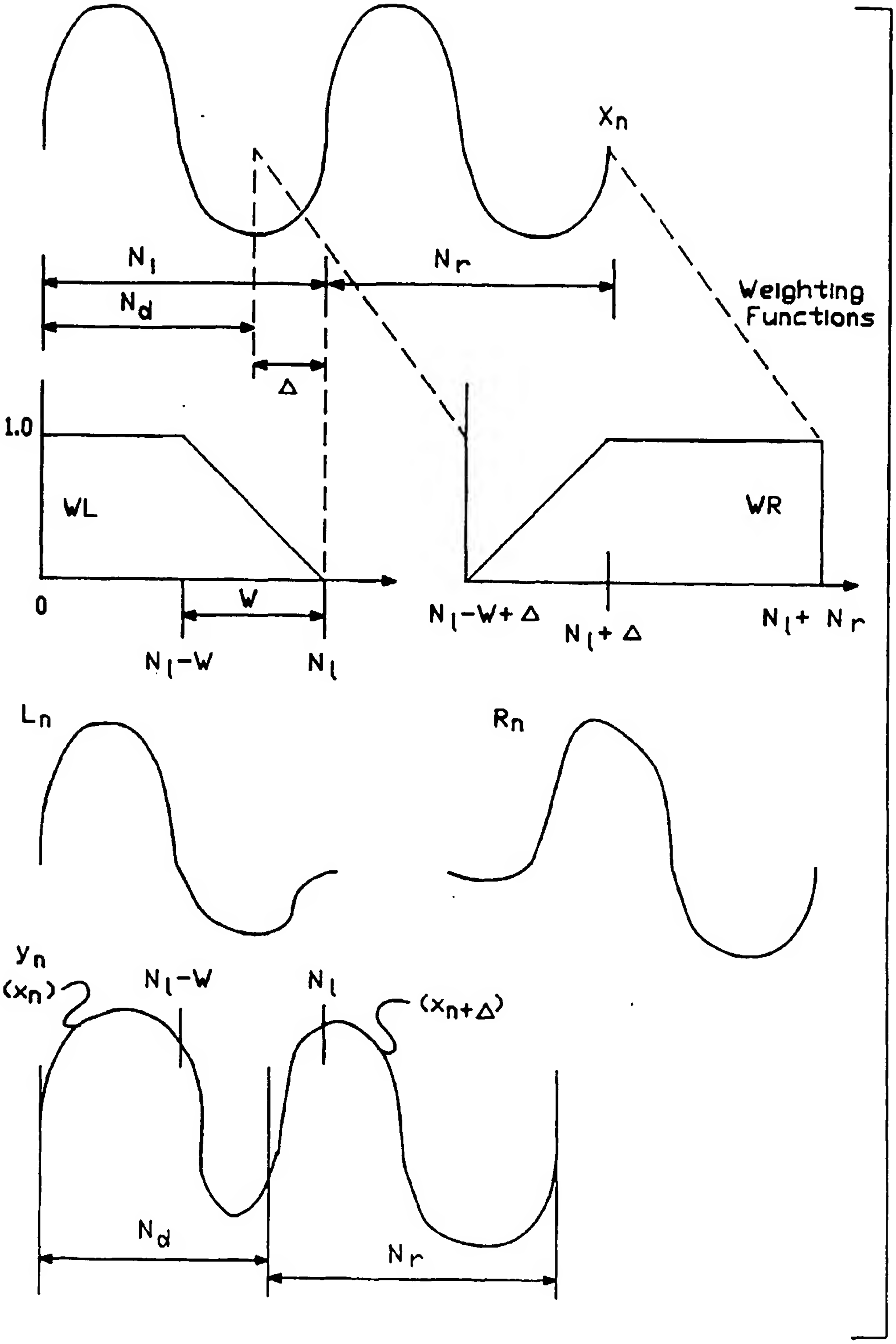

DECREASE PITCH PERIOD TO N - Δ
400
STORE TWO PITCH PERIODS IN BUFFER xn
401
GENERATE LEFT VECTOR
Ln = WL (xn, Nl, W)
(BEGINING MOST SIGNIFICANT)
402
GENERATE RIGHT VECTOR
Ln = WR (xn, Nl + Nr, W)
(ENDING MOST SIGNIFICANT)
403
BLEND Ln AND Rn + Δ
Ln + Rn + Δ
404

FIG.-13



X_n
N_l
N_r
N_d
Δ
Weighting Functions
1.0
WL
WR
0
W
N_l-W
N_l
N_l-W+Δ
N_l+Δ
N_l+ N_r
L_n
R_n
y_n
(x_n)
N_l-W
N_l
($x_{n+Δ}$)
N_d
N_r

FIG.-14

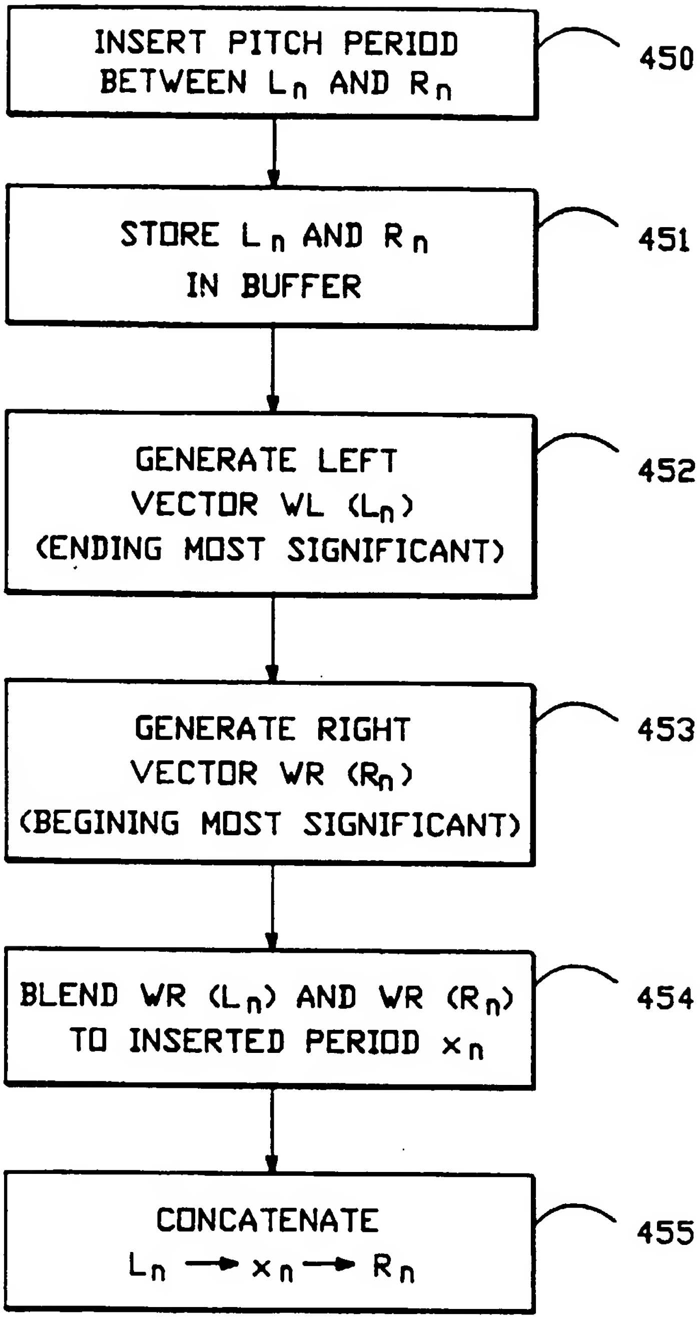

INSERT PITCH PERIOD BETWEEN L_n AND R_n
450
STORE L_n AND R_n IN BUFFER
451
GENERATE LEFT VECTOR WL (L_n) (ENDING MOST SIGNIFICANT)
452
GENERATE RIGHT VECTOR WR (R_n) (BEGINING MOST SIGNIFICANT)
453
BLEND WR (L_n) AND WR (R_n) TO INSERTED PERIOD x_n
454
CONCATENATE $L_n \rightarrow x_n \rightarrow R_n$
455

FIG.−15

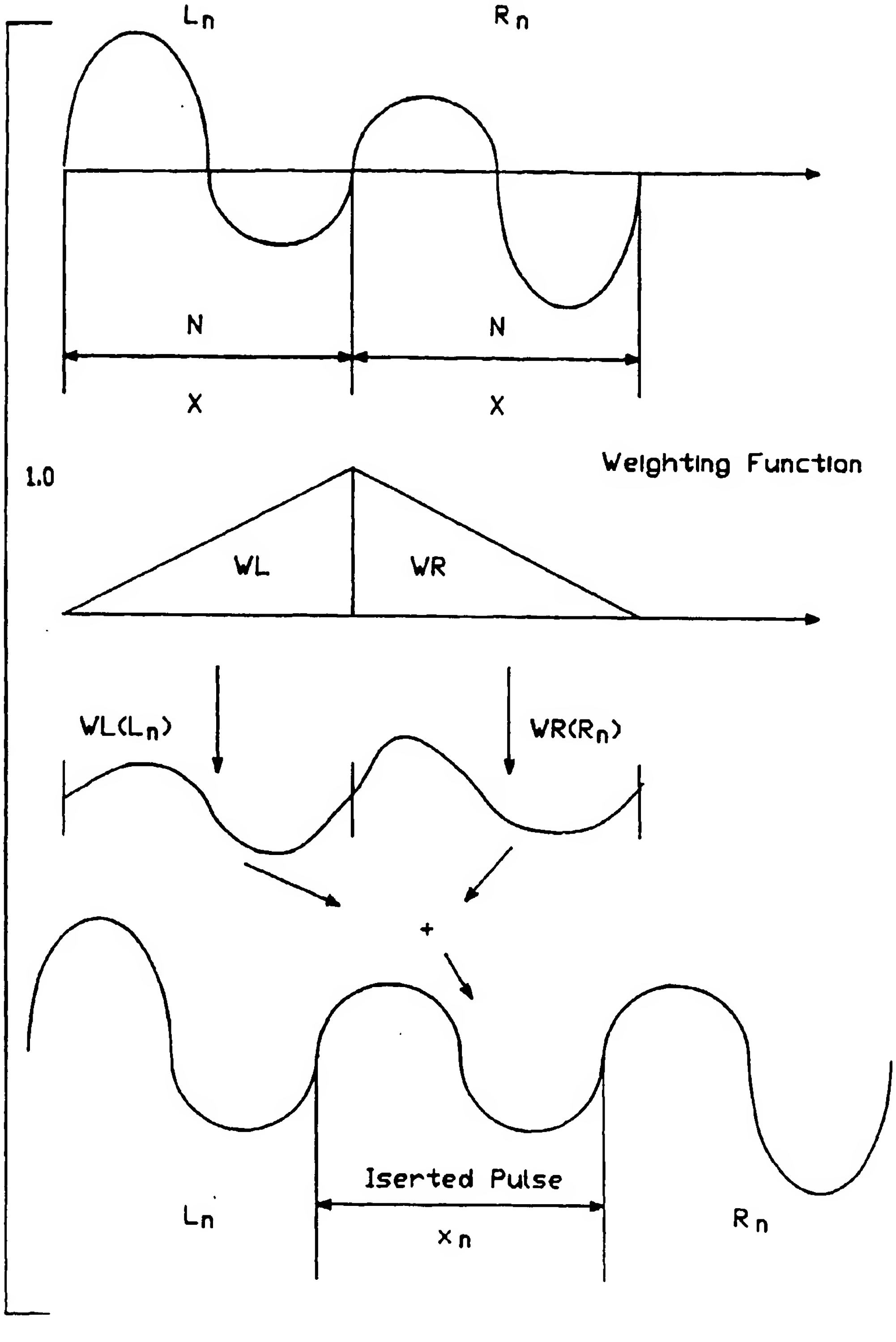

Ln
Rn
N
N
X
X
1.0
Weighting Function
WL
WR
WL(Ln)
WR(Rn)
+
Iserted Pulse
Ln
xn
Rn

FIG.-16

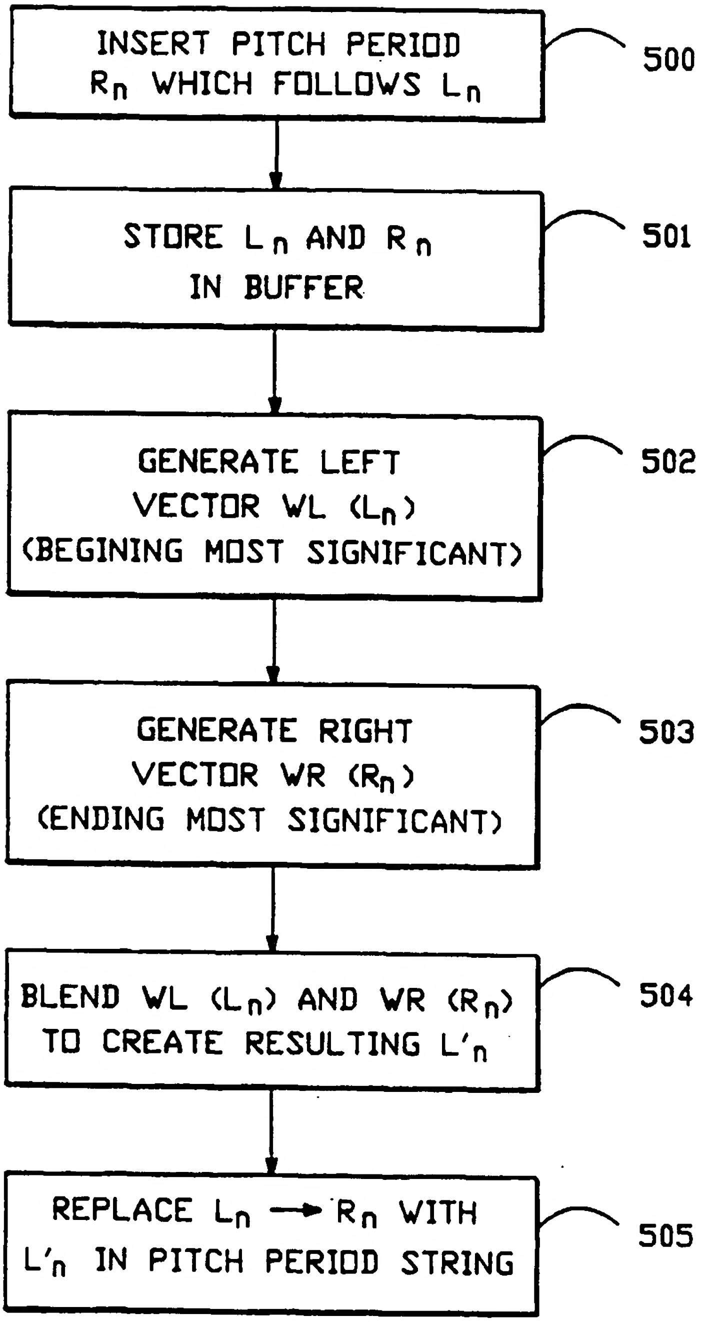
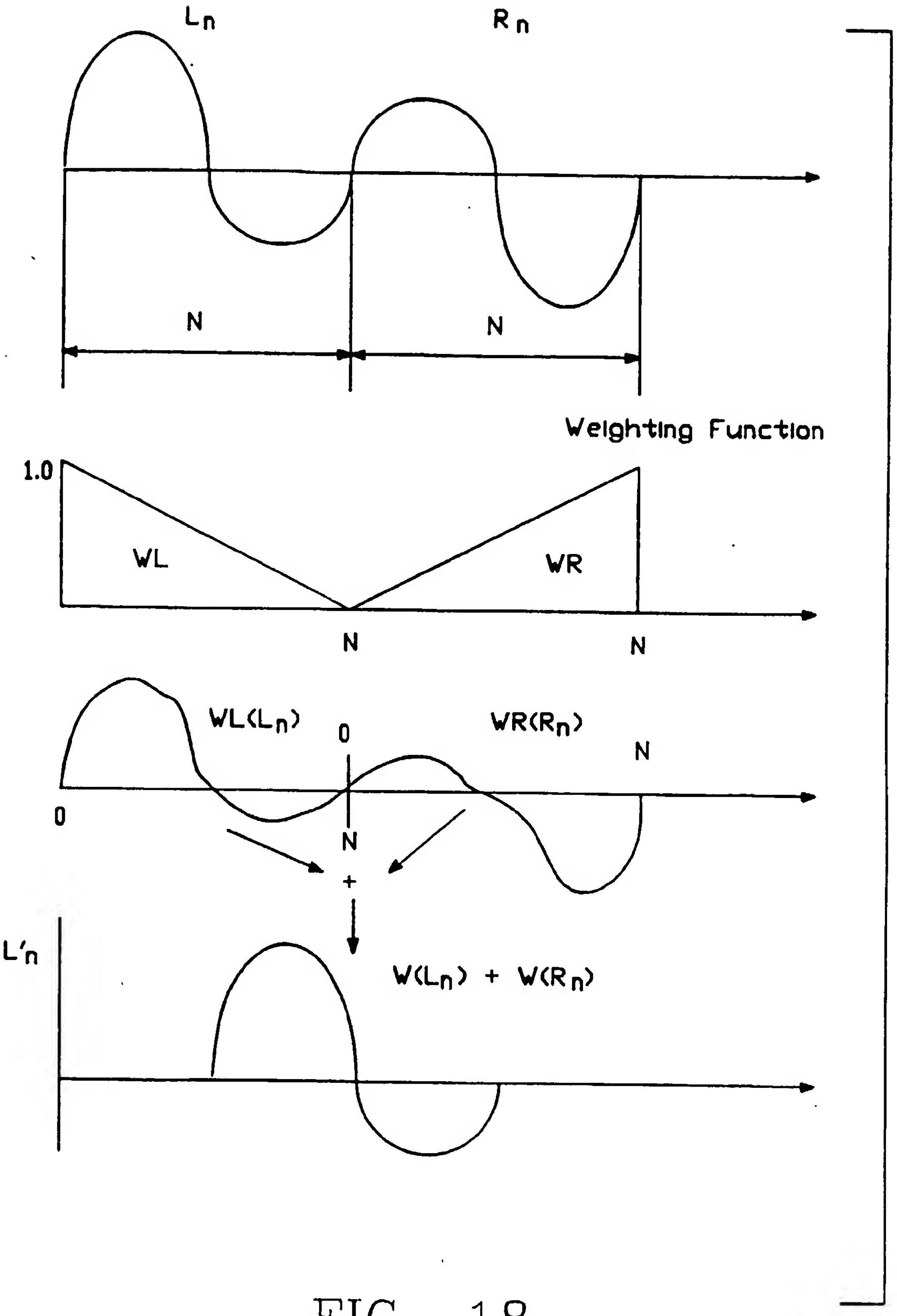

INSERT PITCH PERIOD R_n WHICH FOLLOWS L_n
500
STORE L_n AND R_n IN BUFFER
501
GENERATE LEFT VECTOR WL (L_n) (BEGINING MOST SIGNIFICANT)
502
GENERATE RIGHT VECTOR WR (R_n) (ENDING MOST SIGNIFICANT)
503
BLEND WL (L_n) AND WR (R_n) TO CREATE RESULTING L'_n
504
REPLACE L_n → R_n WITH L'_n IN PITCH PERIOD STRING
505

FIG.−17



L_n
R_n
N
N
Weighting Function
1.0
WL
WR
N
N
WL(L_n)
0
WR(R_n)
N
0
N
+
L'_n
W(L_n) + W(R_n)

FIG.–18